

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Market Update Microphones—What's New? How Big is the Market?

By J. Martins

R&D Stories A Groundbreaking Approach to Gain Control for Conference Calls

By Erica Yorga and Rich Ferguson

Sound Control Positioning Choir Microphones

By Richard Honeycutt

Audio Electronics Powering Your Circuits—Battery Power

By Ed Simon

Fresh From the Bench Vanatoo Transparent One Encore

By Stuart Yaniger

Questions & Answers Designing Speakers An Interview with Charlie Laub

By Shannon Becker

Hollow-State Electronics Effect of Load Impedance on Amplifier Output

By Richard Honeycutt

Audio Scope Interfacing Measurement Mics to Delta-Sigma ADCs

By Michael Steffes





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The International Year of Sound!

2020 is the International Year of Sound (#IYS2020). This is an initiative of the International Commission for Acoustics (ICA) and is an outcome of UNESCO Resolution 39 C/59 2017 "The Importance of Sound in Today's World: Promoting Best Practices" (<https://www.lasemaineduson.org/IMG/pdf/39c-49.engl.pdf>). This follows a proposal approved in the United Nations to highlight "the importance of sound in today's world." The proposal also stated that "the sound environment is a key component in the equilibrium of all human beings in their relationship with others and with the world, in its economic, environmental, societal, medical, industrial and cultural dimensions."

In broad strokes, the resolution states that:

- The sound environment reflects and shapes our individual and collective behavior, and our productivity and capacity to live in harmony together. Giving more importance to sound-related issues in our increasingly noisy world has thus become a vital matter.
- Five sectors are mainly concerned: the sound environment; health; sound recording, reproduction and conservation technology; the relation between image and sound; and musical and sound expression.

I think these very generic but ambitious principles are something that should be in the minds of all *audioXpress* readers and should be highlighted by all companies and organizations in the audio industry in 2020. There are already many workshops, conferences, and events scheduled to take place all year, promoted by the Acoustical Societies from all over the world, including by the Acoustical Society of America, which is one of the sponsors. Much more information is at www.sound2020.org.

And, while I was going through the information available about the multiple #IYS2020 initiatives, I received the book review of *Sensory Evaluation Of Sound*, which is included in this edition. In this review by Richard Honeycutt, the main topic focuses on a field of study that bridges physical measurement and human perception, helping us to improve product design and aural environments.

This is a very extensive work by the author, editor, and researcher, Nick Zacharov, regarding new methods for perceptual evaluation of next generation sound systems. Zacharov is the Lead Technologist at FORCE Technology's SenseLab, and chair of the Radio Communication Sector workgroup that oversees all ITU Recommendations relating to audio quality matters. Those include the ITU-R BS.1534 and ITU-R BS.1116 Recommendations, which are widely adopted throughout the audio industry for things as diverse as the selection of audio codecs to subjective listening tests.

This is related to our fields of interest in *audioXpress*—and certainly extremely relevant during the International Year of Sound—so I couldn't resist diving into the extensive work already presented, published, and available—and which forms the basis for the *Sensory Evaluation Of Sound* book.

There are recent practical results of this work by SenseLab that are extremely important, such as in the field evaluation of audio quality in immersive spatial sound applications and specifically in object-based audio platforms, where the sound reproduction is rendered in the user's environment "whether they are using a 2-channel stereo system (headphone or loudspeaker) or a large number of loudspeaker channels (e.g., 5.1, 10.2, 22.2 channels of loudspeakers)."

These so called Next Generation Audio (NGA) solutions were the target of a recently approved ITU-R recommendation BS.2127: Audio Definition Model renderer for advanced sound systems, which is now being discussed for implementation within other fields, including for live sound and performance applications.

This landmark ITU-R specification for a common worldwide Audio Definition Model (ADM) Renderer creates the foundation for "a new era for sound" and is the result of many years of work, involving more than 5,000 specialists. It is vital for content production and exchange (from audio recordings to television and cinema), using advanced sound systems which combine channel-based and object-based technologies, "whereby the sound field is rendered according to the nature of the loudspeaker configuration (number, loudspeakers location, etc.)," as SenseLab defines. The next step involves new tools for the assessment of audio quality in these "highly immersive spatial sound applications" and defining "novel methods for perceptual evaluation of sound systems."

Certainly much of this work will continue for some time, as the adoption of spatial audio, object-based audio technologies, and distribution platforms will continue to evolve.

J. Martins

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ACO Pacific, Inc.	65	JLI Electronics, Inc.	51
All Electronics Corp.	47	KAB Electro-Acoustics	63
Avel Lindberg, Inc.	37	Linear Integrated Systems	49
AXPONA 2020	59	Menlo Scientific, Ltd.	41
Celestion	43	Parts Express International, Inc.	68
COMSOL, Inc.	11	Profusion plc	2
DPA Microphones	9	Purifi Audio	63
GRAS Sound & Vibration	29	Redco Audio	65
Hammond Manufacturing, Ltd.	3	Schurter, Inc.	47
Hypex Electronics BV	39	Swans Speakers Kits	37
Infineon Technologies AG	15	Triad Magnetics	67
ISE 2020	61		

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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.

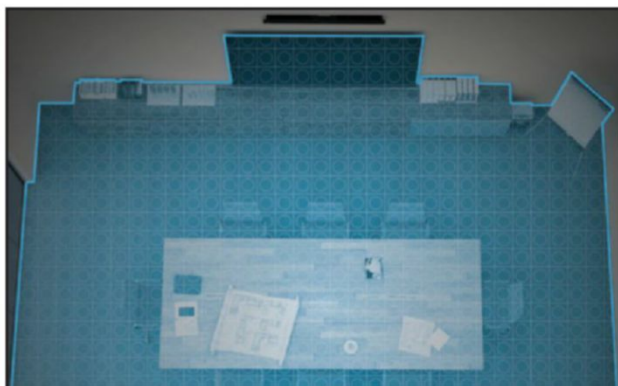
Michael Steffes has worked for more than three decades in high-speed signal path IC development, spanning IC design roles for the BurrBrown high-speed amplifier products, and product launch/application manager roles through Texas Instruments/Intersil. He continues to focus on high-performance ADC interface 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.

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● Features

8 A Look Inside *Sensory Evaluation of Sound*

Review by Richard Honeycutt

Richard Honeycutt reviews *Sensory Evaluation of Sound*, a book edited by Nick Zacharov. This extended work explores the latest sensory evaluation techniques, specifically applied to the evaluation of sound and audio.

12 Exploring *High-Quality Horn Loudspeaker Systems*

Review by Mike Klasco

This book, *High-Quality Horn Loudspeaker Systems—History, Theory and Design*, written by Bjørn Kolbrek and Thomas Dunker, was presented during the 147th Audio Engineering Society (AES) Convention in New York, NY, and is available in a limited first edition. Mike Klasco took the review copy and submitted a “first impressions” review.

30 A Groundbreaking Approach to Gain Control that Cuts Conference Call Annoyance

By Erica Yorga and Rich Ferguson

Nureva, a conferencing and collaboration-solutions company from Calgary, Alberta, was recently granted three new patents for its Microphone Mist technology. *audioXpress* invited Nureva’s engineers to explain the thought process that lead to these innovations.

44 Vanatoo Transparent One Encore

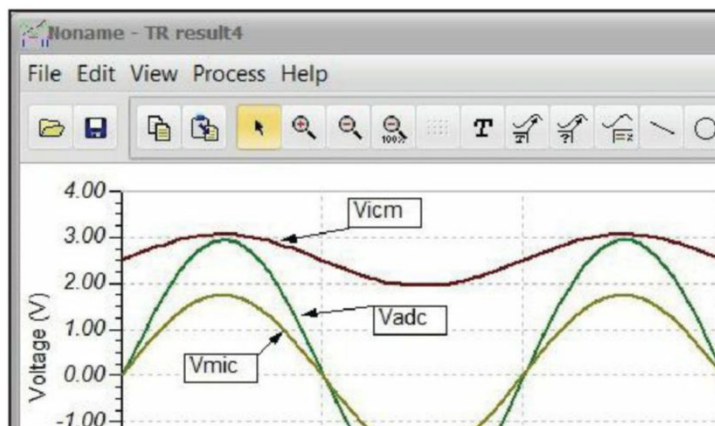
By Stuart Yaniger

Stuart Yaniger discovered Vanatoo’s Transparent Zero a few years ago and he still uses those speakers in his home today. Now, he is putting Vanatoo’s latest offering, the Transparent One Encore (T1E), to the test to see how it stacks up.

56 Powering Your Circuits (Part 1) Battery Power

By Ed Simon

Ed Simon has provided *audioXpress* with a six-part article series that explores the choices and considerations of using different power sources for audio circuits. In this first article, he discusses the positives and negatives of battery power use.



● Market Update—Microphones

16 New and Exciting Microphones

By J. Martins

In this annual Market Update on Microphones, we address the latest technology and application trends in this expanding product segment, which is on the verge of great changes. Following an extremely busy year, we discuss the major trends with stats and product examples.

● Departments

4 From the Editor's Desk

5 Client Index

66 Industry Calendar

● Columns

SOUND CONTROL

34 Positioning Choir Microphones

By Richard Honeycutt

AUDIO SCOPE

40 Interfacing Measurement Mics to Delta-Sigma ADCs Nuances Applying FDAs

By Michael Steffes

QUESTIONS & ANSWERS

52 DIY Expert Enjoys the Challenges of Speaker Design An Interview with Charlie Laub

By Shannon Becker

HOLLOW-STATE ELECTRONICS

60 Effect of Load Impedance on the Output of Hollow-State Power Amplifiers

By Richard Honeycutt

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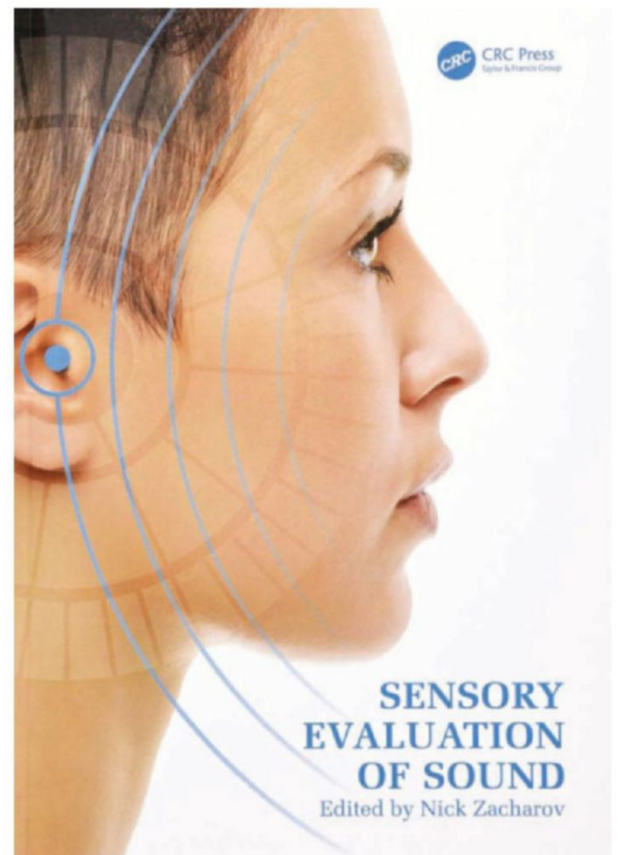
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A Look Inside *Sensory Evaluation of Sound*

Sensory evaluation is a field of study that bridges physical measurement and human perception, helping us to improve product design and aural environments. Richard Honeycutt reviews this extended work, which was edited by Nick Zacharov, exploring the latest sensory evaluation techniques, specifically applied to the evaluation of sound and audio.

Review by

Richard Honeycutt



The field of acoustics is very broad. Many of my clients equate acoustics with sound reinforcement. Others include architectural acoustics. But these sub-fields in themselves include yet other subdivisions: human hearing, perceptual acoustics (also called psychoacoustics), electroacoustics, noise control, and more.

A recent category of interest in acoustics is variously identified as soundscapes, product sound quality, and sensory evaluation of sound. When I first joined the North Carolina regional chapter of the Acoustical Society of America (ASA), I expected to learn a lot about architectural acoustics. I did, but I also learned a good bit about the sub-fields just mentioned, plus animal acoustics and other even less well-known disciplines.

Other than the journals of the various acoustical societies worldwide, there have been few references for many acoustical disciplines. *Sensory Evaluation of Sound*, edited by Nick Zacharov, provides a comprehensive reference on aspects of sound that directly affect us all. The list of contributors to this volume includes many names that are familiar to those in the field of sensory evaluation, but perhaps not

so familiar to the general public. However, the information they supply, and the research they report, are of the highest quality.

The vast contents in the book include an introduction to the field of study that bridges physical measurement and human perception, helping us to improve product design and aural environments. This is followed by several chapters that deserve an explication:

Why sound matters—Sound is employed in human communication and entertainment; but more than that, it surrounds us all the time, affecting our health and mood, as numerous studies have shown.

Sound, hearing, and perception—Sensory evaluation of sound begins with an understanding of sound and audio, including human hearing, psychoacoustics, spatial hearing, human perception of music and speech, and hearing impairment.

Theory and Practice of performing sensory evaluation—Numerous test methods and protocols have been developed for sensory evaluation of sound. The data from testing is usually analyzed using univariate or multivariate statistical methods.

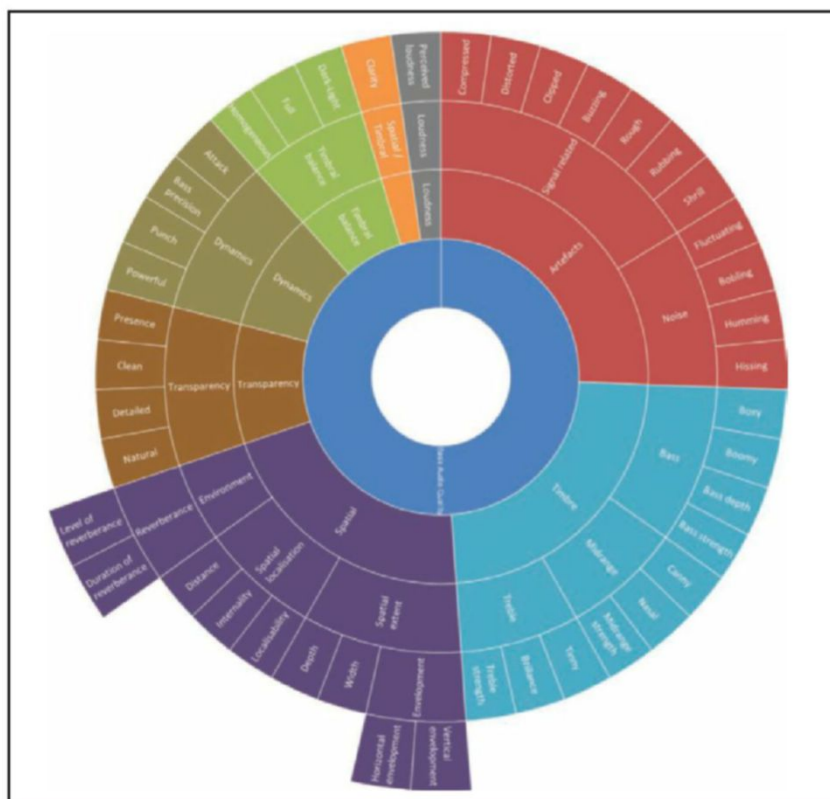
Sensory Evaluation of Sound
Edited by Nick Zacharov
Publisher: CRC Press,
First Edition
December 2019
Hardcover: 578 pages

Applications in telecommunications—Sensory evaluation of sound was initially developed to enhance speech communication via telephones, and continues to be used to enhance speech quality and intelligibility.

Hearing aids—The World Health Organization reports that 5.3% of the world's population is affected by some degree of hearing disability, and a variety of hearing aids have been developed to assist people thus affected. Optimizing hearing aid design involves careful sensory evaluation of product quality and usability.

Car audio—Reproduction of sound in cars involves the acoustics of the vehicle cabin and design of suitable reproducing equipment, including ongoing sensory evaluation of the system performance as perceived by humans.

Binaural spatial reproduction—Despite the ubiquity of loudspeakers in sound reproduction, binaural technology is the approach that most nearly approximates a real-life condition for experiencing sound, since it presents the complete audio signal with spatial information, uncontaminated by the acoustics of the listening space.



The sound quality wheel is an effort to provide a lexicon for generic reproduced sound quality evaluation. The sound attribute wheel has also been contributed to the ITU-R WP6C RG25 for consideration as part of new methods for perceptual audio evaluation. (Audio Engineering Society, 2016)



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Concert hall acoustics—At least since Wallace Sabine's pioneering work on architectural acoustics in the early years of the 20th century, concert-hall sound has been a primary focus of acoustical research. Especially since the middle of that century, sensory evaluation has led to identification of parameters that must be considered in the design and optimization of concert halls.

Emotions, associations, and sound—The emotional effects of sound, particularly music, have been known by all cultures throughout history, but through sensory evaluation studies we have made our understanding more objective, specific, and predictive.

Audiovisual interaction—In an audiovisual presentation such as a movie, the aural and visual components interact, each affecting the other as a unified experience is created for human consciousness. Sensory evaluation is beginning to help us understand this interaction, with the result that higher-quality presentations can be developed.

All these topics are covered in *Sensory Evaluation of Sound*, with enough background to facilitate understanding, although the clear focus of the book is not on the phenomena, but on their evaluation using the human senses.

As Zacharov points out in the foreword, there has been a long-standing disagreement among audio professionals and audiophiles concerning the sufficiency of technical measurements for determining whether an audio system or piece of equipment sounds good. This disagreement is at the heart of discussions concerning the relative merits of tube versus transistor equipment, vinyl versus digital sound recordings, .wav versus MP3 sound-file coding, and so forth.

In the early days of "high fidelity" sound, the usual evaluative approach was to compare the sound of a well-recorded symphony orchestra to that of the recording played on the audio system under investigation. Of course, such an approach is plagued with multiple unavoidable extraneous variables. Nor is it applicable in evaluating either live-sound systems or systems used to reproduce recordings of electrically/electronically created music, for which no acoustical original exists to which the playback can be compared. The science of sensory evaluation provides a set of scientifically grounded approaches to use in the design and comparison of audio systems and other devices that make sound.

There is not enough space in this review to thoroughly discuss all the topics included in this extraordinarily comprehensive tome, but

two examples should suffice. One sub-field of acoustics in which sensory evaluation has often been applied is concert hall acoustics. Although W. C. Sabine is most noted for his pioneering work in identifying, measuring, and predicting the effects of reverberation on an audience's appreciation of speech and music, he also mentioned sufficient loudness, adequately flat frequency response, clarity, and distinctness of articulation, and freedom from noise as essential to good auditorium acoustics.

Over the years, many other attributes of good concert-hall acoustics have been identified, defined, and studied. The resulting research reveals the importance of intimacy, diffusion, balance, blend, ensemble, definition, brilliance, muddiness, hardness, low-frequency balance, high-frequency balance, presence, and envelopment in various known-good and less good halls.

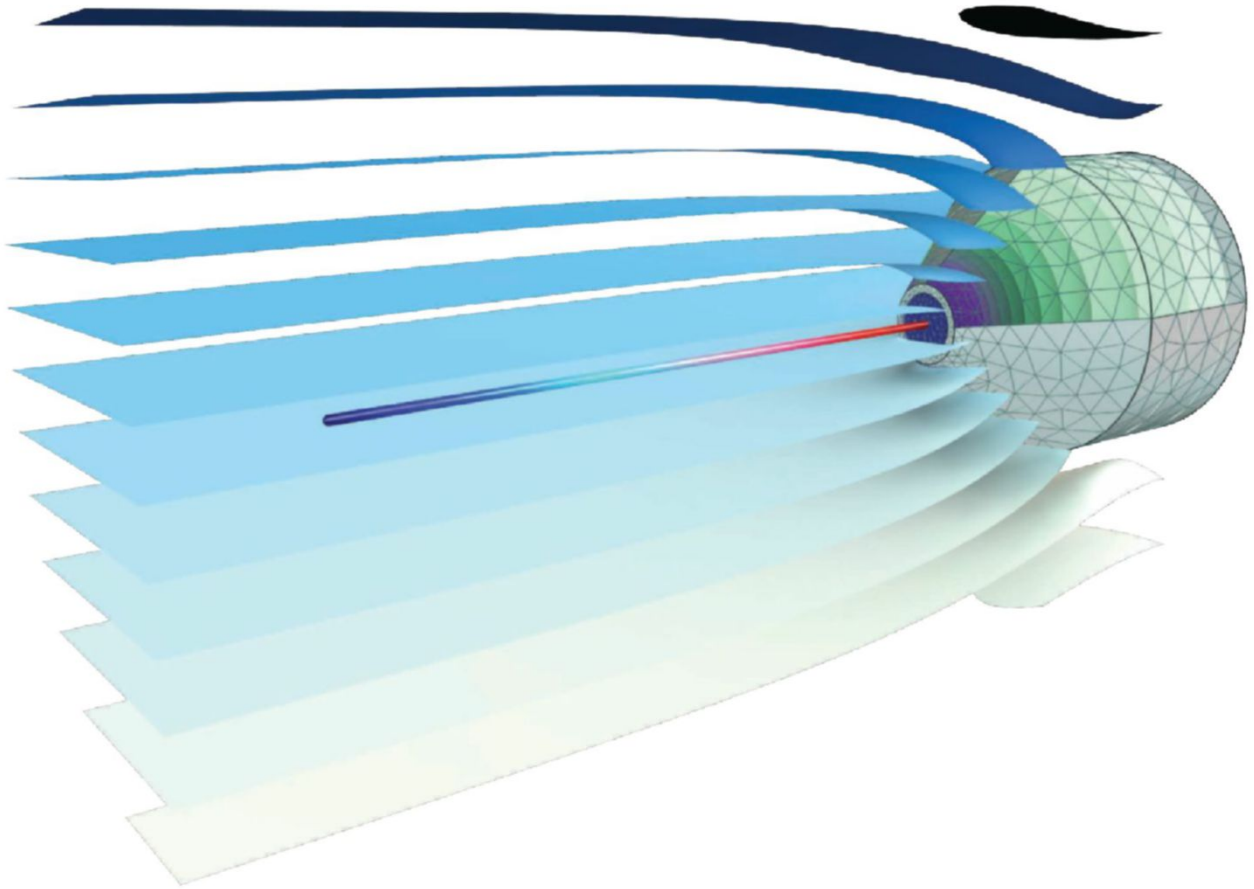
In the ISO standard 3382-1 (2009), the overwhelming list of parameters has been distilled into five: subjective sound level, perceived reverberance, perceived clarity, apparent source width, and envelopment. This codification of parameters important to audience perception is a prime example of careful application of the sensory evaluation methods presented in Zacharov's book.

One important aspect of experiencing a live or recorded musical performance is the evoked emotions. Although nearly everybody recognizes that "music hath charms to soothe a savage beast"—or to calm or excite a human—not many people have any idea that the topic is accessible to scientific study. An excellent discussion of the emotional effect of music can be found on the surprisingly serious video by comedian Bill Bailey at: www.youtube.com/watch?v=AcPTFXzh0LQ. But few print resources are available; the only one that comes to mind is Kay Gardner's *Sounding the Inner Landscape*.

The author of Zacharov's Chapter 13, Jasper Ramsgaard, makes it clear why there is such a dearth of information relating music to emotions. First of all, there is no generally accepted definition of "emotion." To compound the problem, the terms "preferences," "feelings," attitudes," and "moods" are often used as synonyms for "emotions," although their meanings are not the same. By discussing categorical and dimensional models of emotions, direct assessment and self-reporting, and physiological and neuroimaging techniques, this chapter sheds welcome light on this very important topic.

Anyone interested in sound from an amateur or professional perspective will find this book a valuable addition to his or her library. 

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



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

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

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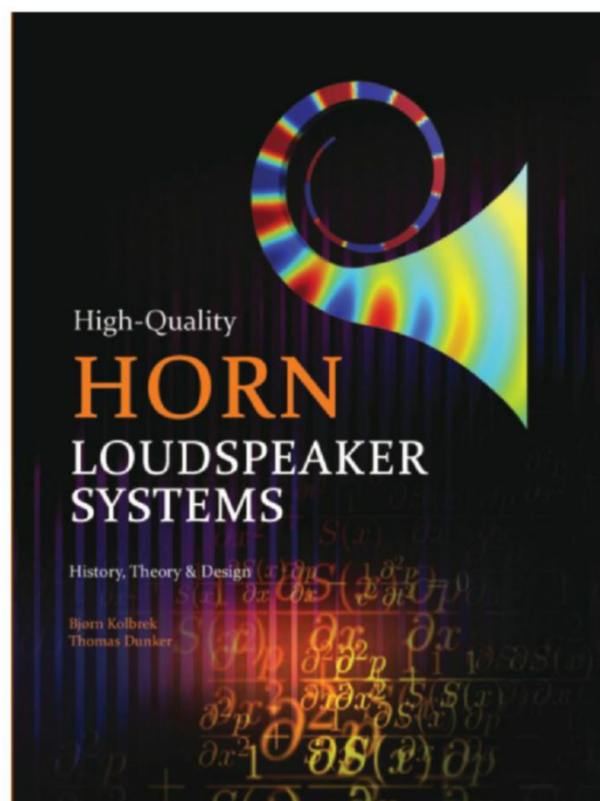
Exploring *High-Quality Horn Loudspeaker Systems*

This book, *High-Quality Horn Loudspeaker Systems—History, Theory and Design*, written by Bjørn Kolbrek and Thomas Dunker, was presented during the 147th Audio Engineering Society (AES) Convention in New York, NY, and is available in a limited first edition. The massive 1,070-page hardcover book caused quite an impression among AES members and *audioXpress* and *Voice Coil* authors. Mike Klasco took the review copy and submitted a “first impressions” review.

Review by

Mike Klasco

(Menlo Scientific, Ltd.)



The first edition of the book is limited to 1,000 copies and will sell for \$150 USD, available only in hardcover, with a quality sewn binding and printed on silk finish paper, 1,070 pages, richly illustrated.

High Quality Horn Loudspeaker Systems
Authors: Bjørn Kolbrek and Thomas Dunker
Publisher: Kolbrek *Elektroakustikk*
First Edition: October 2019
ISBN-10: 152724542X
ISBN-13: 978-1527245426
Hardcover: 1,070 pages

As the title of the book under review clearly states, this is a book about horn history, theory, and design. Development work and papers on horns and compression drivers are well covered in the trade magazines and technical journals. Nevertheless, this topic for the focus of a reference book is a good idea as it has not been addressed lately and has been superbly executed. Co-author Thomas Dunker referred to this book as the missing piece of his library. For speaker engineers, I think we can concur on behalf of all our colleagues. But if I need to be more precise, this is the finest reference work on horn loudspeakers possibly in the history of mankind. I am not worthy to review and judge

this work, but only serve to carry the message of its existence to our readers.

Background Information

The authors are native Norwegians but thankfully for most of us they decided to use their second language, English. While they apologize for any shortcomings in their language skills, their work provides clean, concise, and well-written content resulting in a polished and professional resource book. As a scholarly reference, the table of contents and bibliographies after each chapter are finished off with a full index.

One of the authors, Bjørn Kolbrek, received his doctorate with his dissertation “Extensions to the Mode Matching Method for Horn Loudspeaker Simulation” in 2016. Kolbrek became interested in horn speakers in the late 1990s and was affiliated with Celestion in the UK before finally earning his doctorate at the NTNU in Sweden. Longtime *audioXpress* readers might remember Kolbrek’s “Introduction to Horn Theory Part 1 and Part 2” written back in March and April 2008. But even before this article, the authors collaborated on

horn speaker research in 2004. At that time, both experts were looking for answers to some of the same questions and both had studied the theory of horn speakers for some time and collected a great deal of literature.

Thomas Dunker, the other author, is also a passionate audio experimenter and DIYer, and while horns have remained his avocation rather than vocation, he still has immersed himself in the subject for more than 25 years. The historical aspects found within the book were his main contribution, but Kolbrek also contributed to the history section.

The foreword is by Mark Dodd, which most of us know as a longtime Celestion employee and head of research. The book reflects positively on the entire Celestion team and the brand itself. In recent years, some quite superb and innovative drivers have come from Celestion, so the knowledge gained from this study apparently has “rubbed off” on the product development team!

The Content

But let me push forward into the book itself. *High Quality Horn Loudspeaker Systems* is well beyond comprehensive, it is a truly exhaustive exploration of the field. With apologies to the religiously sensitive, and I say this with all respect—the title would be more accurate if it was *The Bible of High Quality Horn Loudspeaker Systems*. This tome is more than 1,000 pages, the first section is a history of electroacoustic transducers from the telephone receiver to movie sound and broadcast monitors onward. Each pioneering firm’s efforts are documented from Western Electric to AT&T, RCA, Vitavox, and many more.

Most of us know the story that arguably the first speaker was invented by Peter Laurits Jensen, living in Napa, CA. But the real story is that Jensen was not really a local. He was consulting in the area and was still very much Scandinavian (Danish). Aside from the formal history, the authors have distilled many of the “back stories” from the dusty basement archives of several institutions.

With the history out of the way, the book continues onto theory and design with a thorough exploration of horn loudspeakers up through today. While you could read every Audio Engineering Society (AES) paper in its E-library, it is obvious Kolbrek and Dunker beat you to it. Out of curiosity, I picked a couple of obscure topics to see if all the trivia was sorted and covered. Within every nook and cranny I looked, I found the authors had long ago signed the guestbook. In a few instances where I thought something might have been left out, I



Bjørn Kolbrek (right) and Thomas Dunker (left), the authors, are both native Norwegians and collaborated on the book’s extensive research, spanning many years. Kolbrek, a PhD, doctored in 2016 at the NTNU in Trondheim, Norway and works on horn simulation software development at Celestion in the UK. Kolbrek sorts out the math and the heaviest theory and has written the theory sections of the book. Dunker is a passionate audio experimenter and DIYer, and has studied the subject of horn speakers for more than 25 years.

stumbled onto well-researched coverage later on in the book. Of course, as soon as anything is published, on-going developments will continue and I am sure their second edition will be well-received in a couple of years.

Since the pressure was on to publish a review of this book right away, I have yet to completely read the entire 1,070 pages. I did dig deeper on a few sensitive topics such as Constant Directivity. From time to time everyone (and every industry), goes off the rails and chases stray balls. I always felt the pro-sound engineering (and marketing) community’s foray into constant beamwidth as a main performance criteria at the expense of sound quality (transient response and resonance) in the 1970s was a fool’s errand. So I was quite curious how the book would treat what for me was the dark ages of horn design—the attempt at constant directivity (CD) at all cost for a number of years—with only a few speaker engineers not pursuing that avenue of development.



A true treasure trove of information and knowledge, *High-Quality Horn Loudspeaker Systems* opens with 17 pages for the table of contents alone.

Those early efforts resulted in the terrible sounding horns of the 1970s (e.g., the Electro-Voice White Horns, the Altec MantaRay, and a couple of Community CD horns)—they all had throat aspect ratios that heavily compromised clarity in order to gain a wider pattern above 8 KHz. The application of these horns in the cinema resulted in unpleasant and fatiguing dialog. Yet, the mantra for constant directivity was mainly driven by the marketing departments. As expected, the authors’ frank comments about the nasty nasal shortcomings and astute insights on chasing pattern control at the top-end at the expense of acoustic aesthetics hit the nail on the head.

However, not all companies chased the constant directivity mandate, including the Meyer Sound products, some of the Yorkville Pulse series horns, some Beyma designs—these examples found ways to avoid the failings of these early CD designs. Yet, the same design engineers who developed most of the CD horns later went on to give us some very fine horns with extended bandwidth with fairly constant directivity. Don Keele’s bi-radial horn in 1980 managed to deliver both clean sound and good wideband directivity earning JBL renewed respect in studio monitors and the cinema market.

So many topics are addressed that are not commonly discussed, such as the serious

compromises in sound quality that seemingly we all have come to accept. Throat distortion due to higher compression ratios are addressed within a number of sections. Lower compression ratios are a path to enabling horns speakers not to have “horn sound.” Some examples where top-end was intentionally sacrificed (or the driver was intended for three-way designs) such as the Community M4 and the CMR, some of Cliff Hendricksen’s work for Bose Professional such as the L1, the JBL 4” rapid flare throat mid-range compression drivers, and a number of Tannoy co-ax designs.

Similarly, the book also addresses the untidy issue of the expansion rate of the compression driver phase plug not optimally matching to the horn. Yes, it would be a better world if compression drivers were integrated designs as a super tweeter is designed.

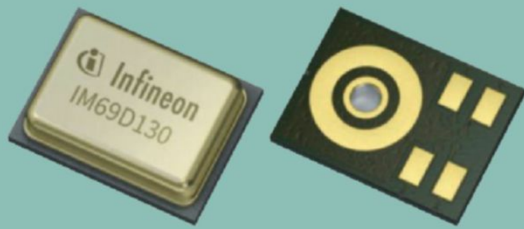
Earl Geddes’ Oblate Spheroidal Waveguide horn is discussed along with how the theory of operation evolved. As a side note, it might be worth a look at the more recent article on this topic that was written by Dario Cinanni and published in the December issue of *Voice Coil* (*audioXpress*’ sister magazine).

Compression driver diaphragms from plastics—polyimide (Kapton)—to and PET (Mylar) are mentioned, along with a discussion of aluminum, titanium, and beryllium but magnesium alloys are not (or it is something I missed). With esoteric materials from beryllium, thin-ply carbon fiber diaphragms (TPCD) and other products now have the promise of being able to lower the compression ratio and not need that path (another pun, sorry) to peak up the highs.

While horn design is a mature field, there are still significant innovations that would have made the early pioneers cheer—from advanced magnetics materials and more recently E.J. Christensen’s air motion transformer (AMT) driver, improving on Oskar Heil’s AMT principle, while it is not covered in the book, the authors do touch upon Eugene (Gene) Czerwinski’s early work on AMT compression drivers at Cerwin-Vega.

Overall Impressions

I will be studying this book for a long time to come and apologize for barely scraping the surface. I am sure, as I continue to immerse myself in this work, it will inspire numerous articles. If your work or hobby has anything to do with horn loudspeakers, you have a duty to acquire *High Quality Horn Loudspeaker Systems*. The audio community owes the authors a thank you for this comprehensive resource.



It is time to be heard – Infineon XENSIV™ MEMS microphones

Define the benchmark in active noise cancellation for a new user experience

The popularity of active noise cancellation headsets is increasing dramatically. However, the signal- to- noise ratio and matching performance of microphones often limits the potential of today's cutting-edge devices.

Not any more!

Our XENSIV™ MEMS microphones introduce a new performance class of digital MEMS microphones that overcomes existing audio chain limitations. Infineon's IM69D130 provides premium digital audio raw data, perfect to combine with advanced audio signal processing algorithms for best noise attenuation experience in any environment.

- › Noise free audio raw data: 69 dB [A] signal to noise ratio
- › Distortion free audio signal capture: <1% total harmonic distortion up to 128 dB SPL
- › Microphone matching: < ±1 dB sensitivity and < ±2° phase response variations
- › Digital PDM interface: ultra-low group delay for latency-critical applications (6 µs at 1 kHz)
- › Wide frequency response: 28 Hz to 20 kHz

New and Exciting Microphones

In this annual Market Update on Microphones, we address the latest technology and application trends in this expanding product segment, which is being heavily disrupted and on the verge of great changes. Following an extremely busy year, we discuss the main trends with stats and product examples.

Antelope Audio is an example of a manufacturer of recording studio electronics that decided to invest in microphone modeling technology, and already offers a complete range of large-diaphragm condensers, with emulations developed specifically for each microphone.

By
J. Martins
(Editor-in-Chief)

In the January 2019 edition of *audioXpress*, we provided a complete overview of the microphone market and trends in our first Market Update feature on Microphones. As we commented then,

while microphones are at the very core of the audio industry in all application segments, seldom do we have the chance to look at this specific category with a complete perspective of the technology itself, and how it is evolving.

The Earthworks SV33 is a front address cardioid condenser microphone that provides the warmth and detail required for the best vocal performances. Incorporating a near-perfect cardioid polar pattern that is consistent up to 80° off-axis, the SV33 allows the singer to move freely in front of the microphone without sacrificing timbre or level. This microphone has very low handling noise and a built-in windscreen to prevent plosives.



As we previously stated, we tend to generalize the microphone market in the same way manufacturers position their products: by application. We have professional audio (stage, music, studio, and broadcast), consumer applications, communications, business (conferencing being the most common) and industrial. This is logical from a manufacturer's commercial organization perspective, much in the same way as the largest microphone brands characterize their own products in terms of end applications, such as microphones for voice (speech or singing), guitar (acoustic or electric), drums (with all the specific varieties from snares and cymbals to the bass drum) to sound recording for film and electronic news gathering (ENG).

Because of this characterization, there is very little data that provides an overview of the actual market in terms of microphone product segments by type of transducer: dynamic, condenser, electret, ribbon, piezo, MEMS, and most importantly, the

size of the diaphragm and its type of interface, considering the electro-acoustical and component level implementation, and the signal response characteristics, of which signal-to-noise ratio (SNR) and maximum sustained SPL level are just some of the key specifications.

This is important because while a professional audio microphone can be characterized by the directional properties, frequency response, and impedance, on the new generation components, those are defined subsequently by the implemented system, considering integrated boosting amplifiers and even digital signal processing.

Likewise, we have the physical interface evolving from simple wired/wireless or analog/digital options, into data interfaces—how the audio signal is transmitted, now including fully networked options with bidirectional data and even power supplied via the network. This enables users to combine a large number of microphone channels and create microphone arrays with very sophisticated digital signal processing possibilities. This evolution from the use of single diaphragms to fully featured microphone arrays enables a world of possible variables, which only recently has started to be fully explored.

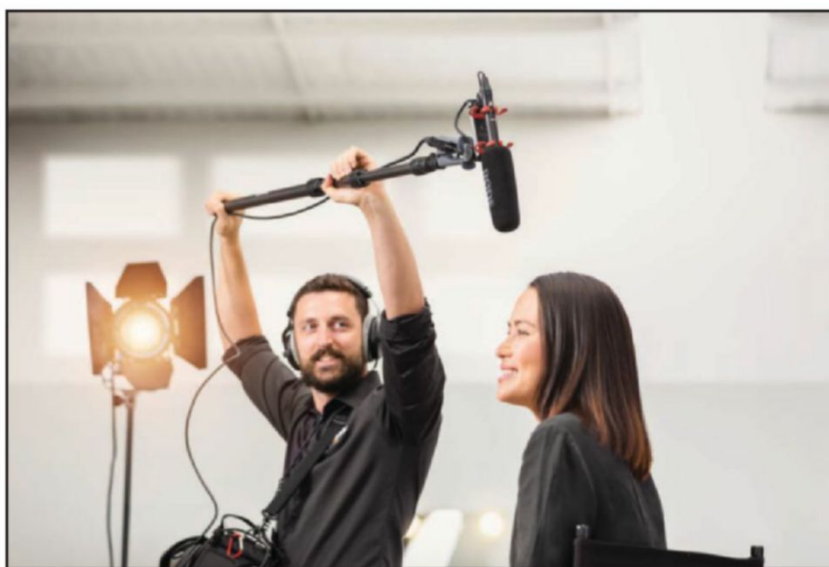
Adding to this, the evolution to very sophisticated digital processing and circuit emulation tools have enabled some clever companies to profile the most popular recording microphones and offer users the ability to instantly switch between hundreds of perfect emulations. The same manufacturers are now able to offer a range of “universal” microphones capsules, which when complemented with simple software plug-ins, allow producers to perfectly emulate the type of sound source or style of production they wish. This trend has been successfully implemented by companies such as Slate Digital, Townsend Labs, Antelope Audio, IK Multimedia, and others, and is paving way for completely new perspectives in the microphone business.

While it's not very likely that consumers would be able to appreciate the appeal of emulating classical microphones and microphone preamps, there's no doubt that the same technologies are certainly going to be explored to process microphone signals in smartphones and could be even be applied to enhance communications or to power new exciting virtual acoustic applications.

In summary, yes, even the traditionally conservative microphone product segment is being heavily disrupted and is on the verge of great changes. For this Market Update, we have focused on the characterization of trends illustrated with practical examples of actual shipping products. 🚚



Audio-Technica introduced these variable coverage U891 UniPoint Boundary Microphones for conference and boardroom installations, with touch-sensitive switches and two-state RGB LED status indicators. These microphones are equipped with an 80 Hz low-cut UniSteep filter to reduce pickup of low-frequency ambient noise, improved immunity from radio frequency interference, and a unique PivotPoint rotating connector that changes the cable exit point without the need for tools. An external contact closure enables users to control a remote device—a camera, for instance—from the switch on the microphone.



Røde Microphones is challenging the leading microphone brands with products such as the supercardioid NTG5, a culmination of three decades of research and development of high-quality RF-bias condenser microphones for broadcast and film applications.

Futuresource Consulting's Professional Microphone Market Report Shows Exciting Growth Opportunities

According to Futuresource's 2019 market report, the professional microphone market continues to display healthy growth and 2019 figures are likely to exceed revenues of \$1.8 billion worldwide. That's an increase of 6% on 2018, with a similar compound annual growth rate (CAGR) out to 2023. Among the growth application areas, the corporate segment continues to create the largest opportunities, followed immediately by the live sound and venue installation verticals.

"Vendors are operating in a polarized market," says James Kirby, Research Analyst at Futuresource Consulting, "with a low-end, high volume, price-led segment acting as a counterpoint to the high-end, feature-rich segment, where customers are looking for quality and willing to pay for it. Somewhere in the middle there's a danger zone and mid-market brands are feeling the squeeze from both ends, as consumers attempt to secure high quality at a low price."

The pro studio vertical is experiencing slow but steady growth, with a projected CAGR way below the market average. Limited market expansion means the majority of revenue is derived from replacement and upgrade activities, which, due to long cycles, is relatively sluggish. However, the pro studio microphone market has a direct effect on home studios, as home studio owners favor the brands used by pro studios, albeit usually choosing to buy less expensive models.

Home studio growth rates are significantly higher than the professional market, with microphone revenues increasing at more than double the rate. The home studio customer base is growing due to a range of influences, as amateurs and professionals alike get involved in creating music, video, and gaming content, while professionals are also moving production out of purpose-built facilities and into their homes.

The live events market, which includes touring bands, concerts and festivals, theater events, sports arenas, concert halls, houses of worship, and karaoke, continues to experience steady growth, with installed applications in theaters and houses of worship leading the growth charge. The karaoke phenomenon is also a significant driver of microphone sales, particularly in Asia and the northern Asian countries of Japan, South Korea, and China. Karaoke accounts for over a quarter of the handheld microphone market, with most karaoke set-ups built around wireless mic technology.

"The corporate segment is the rising star of the professional microphone market," says Kirby, "outperforming all other applications



by some distance. As AV conferencing moves center stage as an essential organizational function, businesses want to be certain that conferencing conversations are clear and make the most of higher bandwidth. They are placing more and better microphones in meeting rooms to make sure people can be heard and to present themselves to customers and prospects in a professional manner. For a long time, visual has been by far the primary focus of meeting-room purchases, but as visual technology penetration reaches record levels, end users are increasingly aware of the high-quality audio systems needed to support seamless AV collaboration."

Futuresource conducted end-user research with corporate AV purchasers and meeting room users in France, Germany, the UK, and the US. A combination of 579 telephone interviews and more than 2,000 web surveys revealed that more than a quarter of companies expect their number of meeting rooms to increase over the next three years, with only a small percentage expecting a decline.

"This expansion is a positive indicator for the growth of conferencing microphones," says Kirby. "Channel and system integrators have confirmed to us that a company's meeting room expansion plans usually provide opportunities to sell microphones. Moreover, our channel contacts report that companies are

increasingly outfitting their meeting rooms with unified communication products and solutions. Over the last five years there's been a marked uptick in ceiling and table-top conference mics for meeting rooms, with beam forming technology, video and audio conferencing, and voice over IP (VoIP) beating a path to better audio and an improved focus on the entire audio signal chain. Combine all this action with more people working from home and it's clear to see the sheer scale of the corporate opportunity."

www.futuresource-consulting.com



The live events market continues to experience steady growth, with installed applications in theaters and houses of worship leading the growth charge. (Photo courtesy of Shure)

Microphones: Show Me The Money!

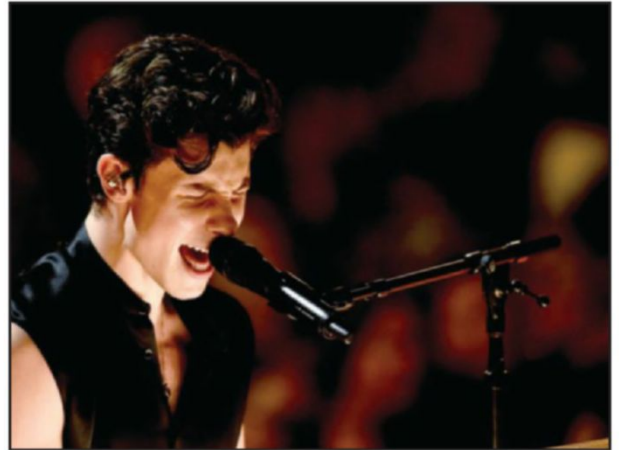
Depending on the perspective and market segment that is analyzed, the statistics available to characterize the overall microphone market continue to be somewhat confusing, even if most of the recent reports agree on a good solid CAGR for the next four to five years.

Providing a broader picture of the market, Futuresource Consulting recently expanded its research into the professional microphone market, which according to the latest 2019 report (see side text, p. 18), continues to display healthy growth. The 2019 figures are forecasted to exceed revenues of \$1.8 billion worldwide. That's an increase of 6% on 2018, with a similar CAGR out to 2023. This can be compared with market research by Arizton, updated in October 2019, which predicts that the professional microphone market is expected to grow at a CAGR of over 2% during the period 2018–2024—a leap of more than \$600 million in revenue between 2015 and 2024. “Buoyed by the demand from the corporate segment and increasing installations in large venues & events, the global shipment of pro microphones will reach more than 5.6 million units by 2024, registering an absolute growth of 37% between 2015 and 2024,” Arizton states. “Although the corporate segment exhibits the highest future revenue projection, vendors could expect major growth traction in the large venues and events segment pertaining to the growing events and live performance market.”

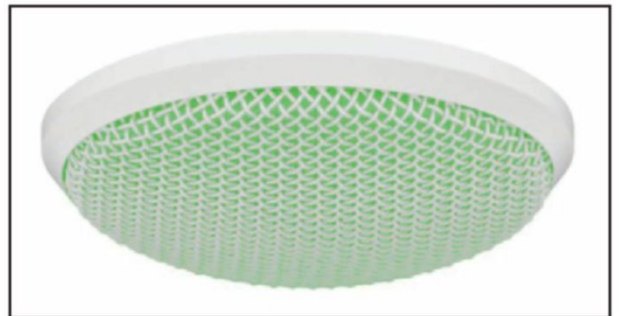
Microphone Components

Other market research on microphone technologies obscures the picture since most of the available reports include component-level products in what's labelled as “professional applications.” That category, which understandably includes microphone capsules used in communications, might effectively include many high-volume microphone capsules used in consumer applications (e.g., gaming headphones). In some cases, it also includes the broad category of microelectromechanical systems (MEMS) microphones, with volumes completely off the scale compared to the overall microphone market. Looking specifically at MEMS microphones is also not easy, since many of the specialized—and more detailed—electronics component market reports throw the category in the more generic “MEMS sensors” banner. As is the case of Yole's latest report, “Microphones, Microspeakers, and Audio Solutions—Market and Technology Trends,” which analyzes cumulatively, the consumer market for microphones, audio IC, and microspeakers and estimates turnover to increase from \$14.1 billion USD in 2018 to \$20.8 billion USD in 2024 with a 6.6% CAGR. “The volume growth of MEMS microphones offers an idea of the products involved. MEMS microphones in smart speakers will experience a 13% CAGR with volumes reaching 1.2 billion units in 2024,” says the market research and strategy consulting company. “In wireless earbuds, MEMS microphones will have a 29% CAGR, reaching 1.3 billion units in 2024,” they add. So, putting this in perspective, while the global professional microphone market represents a \$2 billion USD worldwide industry, MEMS microphone components alone (the dominating product category) could already move much more than that currently, but how much?

Fortunately, there are many companies researching the MEMS microphone market. I found a report from Grand View Research from August 2019 that was particularly detailed and offered very useful data. Analyzing the category by type (digital, analog), by SNR, by technology (capacitive, piezoelectric), and by application, apart from demographic segmentations, this report forecasts the market from 2019-2025.



What people normally associate with microphones is actually a much more complex market, which has never been fully characterized. (Photo courtesy of Sennheiser)



Microphones look different these days. This is the latest Audix M70WD, a flush-mount condenser ceiling microphone designed to be network-connected for Dante/AES67 environments. There's even a plug-in for control and processing via QSC Q-Sys systems.



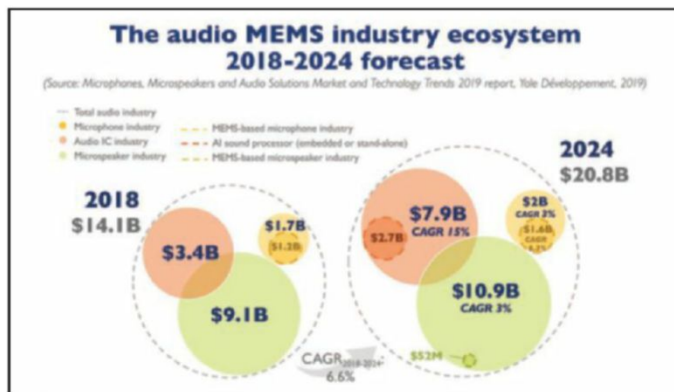
Wireless microphone systems represent more than half of the turnover in professional applications. Sennheiser is among the manufacturers trying to expand the category to the consumer level, with new easy-to-use wireless series that can be used worldwide without a license.

Grand View Research says the global MEMS microphones market size was valued at \$1,207.7 million USD in 2018 and is expected to register a 14.4% CAGR from 2019 to 2025.

Explaining the projected growth, the company forecasts growing demands for smartphones, IoT-enabled devices, hearing aids, and other consumer electronics, among others. “The rising levels of disposable income that have triggered the demand for smartphones are expected to emerge as one of the major factors driving the demand for MEMS microphones. At the same time, the growing aging population, which is suffering from hearing impairment, has triggered the demand for hearing aids featuring superior-quality microphones, which is also expected to drive the market growth over the forecast period.”

MEMS Microphones in Detail

Characterizing the technology trends, which we addressed in our previous Market Update, Grand View Research predicts the emergence of piezoelectric technology—durable, sturdy, power-efficient, and tolerant to dust and moisture—for far-field audio applications, including smart-speakers. But as long as the size of these components remains a constraint, piezoelectric technology



The MEMS microphone and microspeaker ecosystem, according to the French market research firm Yole Développement’s 2019 report on Microphones, Microspeakers, and Audio Solutions—Market and Technology Trends.

Resources

Arizton Advisory & Intelligence, <https://www.arizton.com>

Yole Développement, <http://www.yole.fr>
www.i-micronews.com/products/acoustic-mems-and-audio-solutions-2017

Professional Microphone Market Analysis and Commentary, www.futuresource-consulting.com

Grand View Research, www.grandviewresearch.com/industry-analysis/mems-microphones-market

Zion Market Research, “Microphone Market by Type (MEMS, Electret, and Others) and by Application (Automotive, Medical, Consumer Electronics, Commercial Security & Surveillance, Industrial, and Sensing Application): Global Industry Perspective, Comprehensive Analysis, and Forecast, 2018–2025” Report, <https://www.zionmarketresearch.com/report/microphone-market>

will not be used by the highest-volume categories of smartphones and true wireless earbuds. The rising demand for other consumer electronics is also expected to drive the market growth for MEMS microphones in areas such as tablets, surveillance cameras, home appliances, and service robots.

“Enhancing the overall acoustic impact by canceling any unwanted background noise is very important for these consumer electronics. Hence, microphones used in these consumer electronics need to boast special and unique characteristics. The MEMS technology can play a crucial role toward this end,” says Grand View Research. “MEMS microphones with analog output will continue to be in high-demand—representing approximately 70%-80% of the global market at present—as manufacturers develop less expensive digital output-type microphones. Still, the fact that digital output MEMS microphones are less sensitive to electromagnetic interference and show other key advantages, make it easy to predict that they will become the dominating technology in the future. As such, the digital segment is expected to register a higher growth rate as compared to the analog segment,” confirms the Grand View report.

The need for high SNR MEMS microphones (≥ 64 dB) and miniaturized models that offer similar or better audio quality as compared to electret condenser microphones are a major driver for the industry, with innovation being felt at a fast pace. Component size, SNR, and audio quality are in fact the major arguments that still keep capacitive MEMS microphones in the lead, against the emerging piezoelectric designs, which are currently favored for applications in the harshest environments. And one the harshest environments for electronics also represents one of the largest growth opportunities for MEMS microphone applications in the near-future: automotive. Currently used for beamforming, active noise cancellation, infotainment systems, and diagnosis of engine problems for the prevention of potential crashes, as Grand View Research highlights, these components will be vital also in navigation systems for autonomous driving, and even for environment awareness and mandatory presence-signaling in fully electric vehicles.

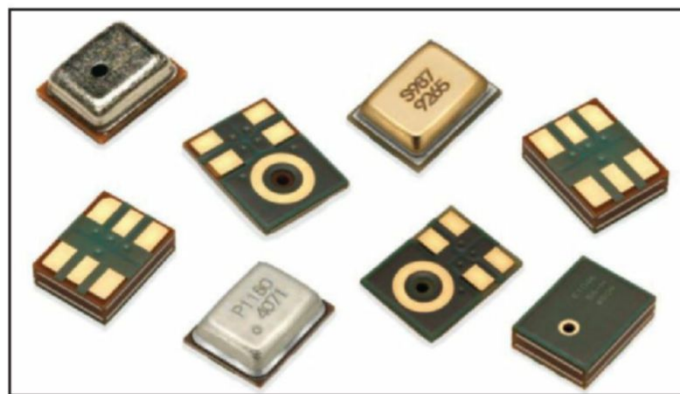
While demand for MEMS microphones will be felt globally, the prices of MEMS microphones are expected to decline over the forecast period, eroding profit margins for manufacturers. As Grand View Research points out, this will no doubt contribute to further industrial consolidation and greater focus on investing in research and development activities and industry partnerships.

Do the Math

Taking all this into account, what is the size of the overall global microphone market after all? Considering that the microphone market includes finished products and component-level businesses; that finished products, like professional wireless microphone systems, actually include more electronics and RF systems than actual “microphones;” and that consumer products that use microphones are only accountable by looking at the actual components sold; there’s no easy answer. But even taking into account that the microphone market entails two completely different types of business, how much is the actual microphone industry actually worth, globally? A dubious recent market report entitled “Global Microphones Market Growth 2019-2024,” whose reliability is as uncertain as its provenance (any

attempt to identify the legitimate source proved impossible) states that the microphone market—including wireless and wired microphones, represented revenues of \$2,820 million USD in 2019 and will reach \$3,460 million USD by 2024, a 4.2% CAGR. Wireless microphone representing 55.05% of global sales volume, according to this study, of which we know very little. Another more conservative but detailed study, this time from Zion Market Research, a firm that actually can be found and contacted, says that “the global microphone market was valued at around \$1,475 million USD in 2018 and is expected to reach approximately \$2,388 million USD by 2025, at a CAGR of around 7.06% between 2019 and 2025.” The Zion report, published in June 2019, and titled “Global Microphone Market: Size, Share, Trends, Segments, Analysis and Forecast, 2018-2025,” actually says that it considers ALL product types (MEMS, electret, and others).

With so many contradictions, it is absolutely impossible to work with these stats. After all, if the professional microphone market alone is worth \$1.8 billion worldwide (according to Futuresource); if Sennheiser and Shure (two of the market leaders that only make microphones and headphones, professional and consumer) have annual revenues



Knowles SiSonic surface-mount MEMS microphones continue to expand in their fourth generation of development and are offered in both analog and digital versions.

of approximately \$700 and \$500 million USD, respectively; and that MEMS microphones alone represented \$1.1 billion in revenue, in 2018 (according to Yole Développement), it becomes obvious that the global microphone market was already worth more than \$3 billion in 2018, and it is probably closer to representing \$6 billion USD, considering all the professional and consumer finished products and component categories. For whatever that is worth. 

Harman Professional Solutions AKG Lyra Multimode USB Microphone

Harman Professional Solutions recently launched the AKG Lyra multi-mode USB microphone, a completely new concept designed to be used in the studio for music recording but also for podcasts and consumer recordings (e.g., YouTube videos). Featuring advanced circuitry that supports 24-bit/192 kHz audio resolution, the AKG Lyra uses a new AKG Adaptive Capsule Array that provides user-selectable capture modes optimized for any application. Everything in the new AKG Lyra USB microphone was designed to make it an easy-to-use solution. An internal self-adjusting shock mount minimizes contact noise, while a built-in sound diffuser and AKG’s proprietary Internal Element Overload Prevention automatically reduces noise, eliminates pops and improves signal levels.

The AKG Adaptive Capsule Array uses four versatile capture modes to adapt to performance needs, eliminating guesswork and enabling users to capture audio in a wide range of recording scenarios. Front mode isolates the target sound from distracting background noise by rejecting unwanted sounds at the microphone’s back and sides, perfect for capturing or live-streaming spoken word, vocals, or close-mic’d instruments. Front & Back mode captures and blends together sound equally on all sides, which is ideal for face-to-face interviews and duets. Tight Stereo mode captures audio in true stereo with discrete left and right audio, providing great separation for side-by-side interviews or panel discussions, or for recording instruments (e.g., drums or piano). Wide Stereo mode captures audio with

greater stereo separation, room ambience, and depth. This is ideal for dynamic situations where users want to capture sounds from all around the mic such as documentaries, field recordings, or spontaneous jam sessions. With no assembly, no need for a separate audio interface and seamless software integration, Lyra works right out of the box and is compatible with Windows, Mac, iOS, and Android devices. The microphone is equipped with a “zero-latency” headphone jack and headphone volume knob that eliminates the short delay between speaking into a microphone and hearing the signal in the headphones, so creators can stay focused on their performance.

www.akg.com/lyra



Inspired by classic studio microphones such as the legendary AKG C414, the new Lyra USB features a sleek design that combines fast and easy controls, including an integrated desktop stand.



The AKG Adaptive Capsule Array uses four versatile capture modes, enabling users to capture audio in a wide range of recording scenarios.

Austrian Audio OC818 Variable Polar Pattern Microphone

Reborn from the ashes of the closing of AKG in Austria, Austrian Audio is the new company that promises “Made in Vienna. Again!” After two years working in stealth mode, while the company created completely new R&D facilities with its own proprietary instrumentation and production machines, Austrian Audio is now in the market with high-quality microphones and headphones.

Austrian Audio’s first product, the OC818 large diaphragm microphone, not only features multiple polar patterns, but also offers innovative remote control possibilities, using the same CKR12 ceramic

capsule as the OC18 microphone, which is a fixed cardioid version.

Handmade in Vienna, the OC818 is a multi-pattern dual-output condenser microphone equipped with a world-first: optional wireless control. The OC818 also features a second output that allows the user to record the rear-facing capsule independently of the forward-facing capsule. If both capsules are recorded to a stereo or dual-mono track, the company’s free and open source PolarDesigner plug-in (Mac/Win: VST, AU & AAX) allows unprecedented control and tailoring of polar patterns between one and five bands with selectable crossover points.

At the heart of the OC818 is the patent-pending, handmade CKR12 ceramic capsule, made to the same critical dimensions as the best legacy CK12 capsules but with far more stability which means every OC818 is inherently paired with any other OC818, or even the OC18, its smaller and more affordable sibling. The multiple polar pattern switching is implemented via microprocessor control of polarization voltages, allowing its activation wirelessly, as well as choosing between two different analog high-pass filters with three settings and two different types of analog pads. Dual outputs allow users to separately record each diaphragm.

www.austrian.audio



DPA Microphones 4097 CORE Supercardioid Choir Microphone

Leading Danish manufacturer of high-quality condenser microphone solutions for professional applications, DPA Microphones continues to expand its technology leadership in extremely high-quality miniature microphones since the introduction of its CORE technology. Already available in d:screet miniature lavalier, d:fine headset microphones, and now also in the d:vote instrument microphones series, basically, the CORE by DPA amplifier technology minimizes distortion and increases the dynamic range of all its product lines, as much as up to 14 dB depending on the specific type of capsule. The CORE microphones are also IP58 certified thanks to the water-repellant nano-coating of the cover and the housing, hermetic sealing of the sensitive amplifier and dual gold plating of the diaphragm. In 2018, DPA launched the new DPA d:screet CORE 6060 and 6061 Subminiature Microphones with just 3 mm in diameter—2 mm smaller than DPA’s existing 4000 series of miniature microphone products. And even with the reduction in size, the Danish company stated these were its best microphones ever made. Proving once again what this R&D effort enables, DPA recently

launched the 4097 CORE Supercardioid Choir Microphone, designed to be one of the most natural-sounding solutions available. This new model provides the same sonic qualities as the brand’s d:vote CORE 4099 Instrument Microphone but was designed specifically to capture dynamic choir sound, with both wireless or wired configurations, featuring a flat off-axis supercardioid pattern. This microphone has a very high-gain-before-feedback, and due to the off-axis characteristic for which DPA is renowned, it makes the entire choir sound natural—not just on axis—but from all

angles. This makes using multiple mics on the choir easy for the sound engineer as he can easily mix the various parts of the choir and blend it all together without having to fight the artifacts created from uneven mic pattern pick-up. Configured with a MicroDot connector and a DPA adapter for wired or wireless, the 4097 enables the user to go wireless, which allows for great flexibility on stage when there are multiple uses of the same space. And the CORE 4099 also includes three shock mounts to ensure that rumble caused by movement on the stage is damped effectively, and at a level not seen before in these kind of microphones.

www.dpamicrophones.com



DPA Microphones Challenge the Vocal Microphone Live Performance Standard

After a considerable period of beta testing and real-world trial, DPA Microphones introduced the 2028 Vocal Microphone, designed to excel in live performance environments and survive the road. The 2028 will compete in a crowded and traditionally conservative market segment by combining a competitive price and DPA's renowned natural sound, which allows all types of vocals to shine.

The Danish manufacturer designed the 2028 to offer an experience as close as possible to the brand's flagship handheld mic, the 4018 VL. It has also been optimized, like the 4018, for the unique challenges of the live stage. "We have seen a need for an excellent vocal microphone that has the sonic qualities of our d:facto 4018 Series, but at a lower price-point," says René Mørch, product manager, DPA Microphones. "At only \$700 USD, this mic is an overachiever—it performs significantly better than anything

else in its price range."

The 2028 features a brand-new fixed-position capsule, as well as a specially designed shock-mount and pop filter. It exhibits a supercardioid polar pattern, with the famous DPA uniform off-axis response. This gives the microphone a very high-gain-before-feedback and makes it easier to handle bleed from other instruments in close proximity, picking up sound in a natural way. "We've designed this mic so that it sounds like the singer isn't actually using a mic," adds Mørch. "You get the full, natural sound of the artist's voice, not what the microphone 'thinks' the artist sounds like. This gives the sound engineer a lot of freedom to be creative—and instead work on crafting a unique sound for the performance based on a clean, natural vocal track."

With the expected wear-and-tear that comes with live performances, both the outer grille and the inner pop filter of the 2028 can be detached and rinsed. The 2028 is offered in three variants; a wired XLR with handle and two wireless mic configurations that are compatible with the industry's most widely used wireless microphone handheld transmitters. This includes the SL1 adapter, compatible with Shure, Sony, and Lectrosonics; and the SE2 adapter, compatible with Sennheiser transmitters.

www.dpamicrophones.com



The DPA 2028 microphone and its capsule shown in detail (at right).



Earthworks SR314—A Bold New Vocal Microphone

Released in March 2019, the Earthworks SR314 delivered a boldly designed vocal microphone to the market with its precision-machined stainless steel chassis and retro-meets-futuristic design. Entering one of the most competitive spaces in the microphone market, the new SR314 from the New Hampshire-based company is an attempt to refresh the live music space that is currently fueling the industry.

The Earthworks SR314 is a precision-engineered cardioid condenser microphone designed from the ground up for live performance environments, and features a rich open natural sound captured in a tight cardioid polar pattern that is consistent throughout the extended 20 Hz to 30 kHz frequency range, with 145 dB SPL handling. The SR314 is now available in three eye-catching combinations—stainless steel, black coated stainless steel with a black windscreen, and black coated stainless steel with a stainless windscreen. It ships with an MC4 microphone clip and 8.5" padded protective bag for \$699 (MAP).

www.earthworksaudio.com



Sennheiser Microphones for Presentation and Education Solutions

Directly investing in the highest growth application markets, Sennheiser has significantly expanded its range of solutions addressing the presentation, conferencing, and educational markets, offering an innovative combination of hardware and software products. Highlighting Sennheiser's business and education solutions portfolio is the new TeamConnect Ceiling 2 (TCC2) microphone array, which uses its patented automatic, dynamic beamforming technology to deliver improved virtual meeting experiences.

Shipping since April 2019, this flagship conferencing solution follows the active speaker in the room and, unlike ceiling microphones with static beamforming technology that work with predefined speaking zones, the improved TeamConnect Ceiling 2 doesn't need to be readjusted when the number of participants changes or when a different seating arrangement is chosen. Due to its single, flexible beam technology, the TCC2 microphone array can be adapted to any type of meeting room and can be used together with existing analog conference systems, or integrated into fully digital systems, controlled and monitored over the network. The TCC2 supports Power over Ethernet (PoE), Sennheiser Control Cockpit software, Dante Domain Manager and provides an open media protocol for third-party control platforms such as Crestron.

The latest version of Sennheiser's Control Cockpit software offers an intuitive, easy way to set up and manage TeamConnect Ceiling 2 along with the SpeechLine Digital Wireless systems, and is accessible in web browsers on any smartphone, tablet, or notebook. It offers a continuous, global dashboard overview of all network-enabled Sennheiser devices, while greatly simplifying adjusting

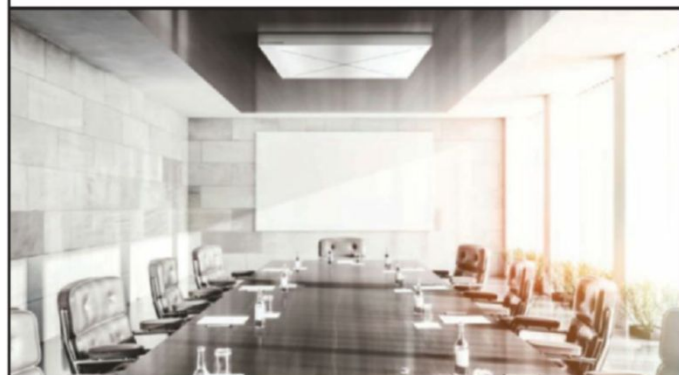
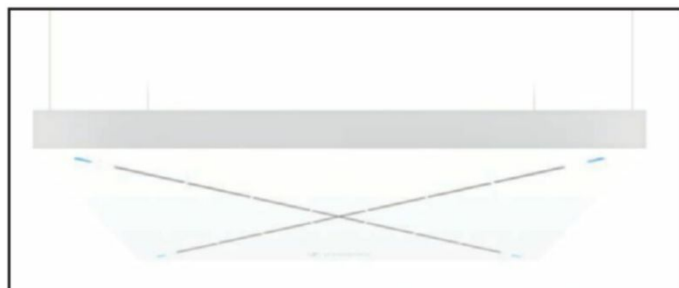
settings for individual or multiple devices. The software provides an overview of coverage field, allows selecting exclusion zones for TeamConnect Ceiling 2 (e.g., to minimize noise from projectors or air-conditioning units), and alerts supervisors when a wireless microphone leaves the range of the receiver.

SpeechLine Digital Wireless and MobileConnect

Created specifically for speech applications in corporate and educational environments (e.g., universities), Sennheiser's SpeechLine Digital Wireless microphone line provides not only the highest level of speech intelligibility and sound quality but now also offers integration with assistive listening solutions.



Sennheiser's SpeechLine Digital Wireless microphone series offers solutions that are specially customized for speech applications. The Control Cockpit (center iPad) application allows for a controlling and monitoring company or campus-wide installations.



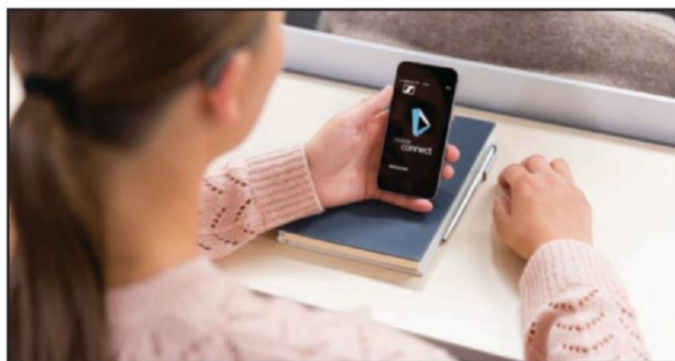
Sennheiser's TeamConnect Ceiling 2 is a conferencing solution that employs automatic, adaptive beamforming technology.



A handheld transmitter—or a bodypack transmitter with a clip-on or a headset microphone—enables the presenter to move freely.

A completely scalable system that is perfect for company/campus-wide installations, the new SpeechLine Digital Wireless systems deliver flexible configuration options and can be connected with MobileConnect, a Wi-Fi-based system for accessible hearing. The combination is ideal for applications in universities and employs a bring-your-own-device (BYOD) philosophy so students can connect using their own device to participate in lectures with personalized audio that ensures greater intelligibility. MobileConnect also supports applications for use in theater, opera, and musical environments.

www.sennheiser.com



Sennheiser Mobile Connect provides direct hearing support via the user's smartphone, streaming audio content live via Wi-Fi.

Townsend Labs Sphere L22 and Dozens of Microphones

Since its debut in 2017, the Townsend Labs Sphere L22 has established itself as a credible microphone modeling solution, thanks also to its pioneering multi-dimensional modeling technology and unique dual-channel design, taking microphone modeling far beyond merely mimicking the static on-axis response of the original microphones. Following more than three years of technical development, Townsend Labs was founded in 2015 by DSP architect Chris Townsend, a former Avid engineer, together with Erik Papp, previously CEO of Summit Audio.

The company, based in the San Francisco Bay Area of California, unveiled its world-class patented microphone modeling technology, Sphere, in July 2016. The announcement was followed by an extremely successful crowdfunding campaign a month later. Sphere's analog hardware directly captures spatial and distance information in the sound field, allowing for unprecedented control and multi-dimensional modeling of vintage microphones from within the Sphere software plug-in.

The Sphere L22 microphone employs a custom designed, dual-channel large-diaphragm condenser capsule that captures the spatial in-room behavior of the original microphones, including transient response, harmonics, proximity effect, and 3D polar response. With just one Sphere L22 microphone it's possible to make coincident stereo recordings, and even have different mic models on the left and right channels, using the Sphere 180 plug-in. The Sphere plug-in is available in all major native plug-in formats, as well as Universal Audio's UAD-2/Apollo and Avid's AAX DSP platforms. It already includes more than 30 models of coveted microphones, and new microphone models are regularly added.

On the company's website, users can explore the system and download various sound examples recorded with Sphere L22 microphones and try the free, fully-featured Sphere plug-in on their system. The Townsend Labs concept excites users' imagination by offering not only exact emulations of the most popular microphones, but also complete sets like the Ocean Way Microphone Collection with 12 ultra-precise emulations of the vintage mics featured in Allen Sides' world-renowned mic locker, or the Bill Putnam Mic Collection, with exacting emulations of priceless vintage microphones used to record Frank Sinatra, Bing Crosby, the Beach Boys, and countless others. With each microphone model, users can freely experiment with different polar patterns, instantly compare

between different versions of the same microphone, adjust the proximity effect, or even use the Dual Mode feature, which enables phase-coherent mixing of multiple models from the same physical microphone, before or after tracking. It's even possible to adjust the relative phase alignment of the two microphones with the Align control. This is not only a great creative tool, but something of immense value for training and education.

www.townsendlabs.com



One microphone that is able to emulate the unique features of dozens of different microphones, digitally processed with very low latency.



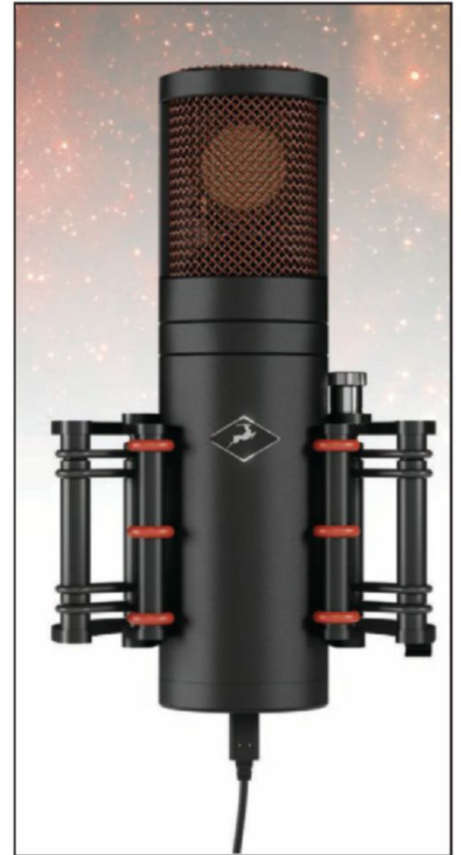
The Sphere L22 microphone modeling system comprises a precision dual-channel microphone, plus an audio plug-in that does all of the DSP processing used to create the microphone models.

Antelope Audio World's First USB-C Bus-Powered Modeling Mic

Professional audio manufacturer Antelope Audio is firing on all cylinders with industry-leading products for modern recording studios. Introduced in 2019, and already in its third generation of microphone modeling solutions, Edge Go is the world's first USB-C bus-powered modeling microphone, enabling a complete studio experience with circuit-level mic and gear emulations onboard Antelope Audio's real-time FPGA FX platform. While professional audio engineers tend to dismiss USB microphones as cheap, unimpressive products for beginners—and in many cases, they have a point—Antelope thinks it is possible to have the full, non-compromised fidelity of the Edge Duo multi-pattern large-diaphragm condenser mic, powered by a console-grade mic preamp, high-end AD converter, and signature Antelope Audio FPGA FX processing, all packed into a premium bus-powered mic, leveraging the increased power of an USB-C interface.

Antelope Audio's Edge Family of modeling microphones' newest member, the Edge Go is a complete recording studio experience housed in a mic. Connected via USB-C and equipped with a 3.5 mm headphone jack for real-time monitoring, the microphone delivers an entire processed vocal chain at the hardware level—and all in real-time. The solution offers access to a variety of accurate emulations of several high-end condenser mic models and a special selection of lifelike vintage compressor and EQ effects, plus built-in de-essing, gating/expansion, tape saturation, and reverb, all controlled by an intuitive PC/Mac app. This makes it even easier to quickly achieve a sound that's ready for radio, podcast, or an album. And it all comes straight off a large-diaphragm condenser microphone, with practically imperceptible latency. The USB-C powered microphone also offers switchable polar patterns (cardioid, figure 8, omni, and everything in between), onboard effects processing and 24-bit/192 kHz A/D conversion.

www.antelopeaudio.com



Røde Microphones Lavalier GO Professional Microphone

Challenging well-established leaders in this space, Australian brand Røde Microphones launched Lavalier GO, a high-quality, professional-grade lavalier microphone designed to pair perfectly with the company's Wireless GO transmitter and most recording devices with a 3.5 mm TRS microphone input. The

ultra-compact and light Wireless GO transmitter is by itself a remarkable new design. The transmitter acts as either a quick and easy clip-on wireless microphone, thanks to a high-quality built-in omnidirectional condenser capsule, or as the world's smallest belt-pack transmitter for the Lavalier GO or any lavalier microphone via the 3.5 mm TRS input.

For users who want the extra flexibility of a lapel mic, the Lavalier GO is the perfect solution either for presentation or interviews. The new miniature microphone offers a permanently polarized condenser omni-directional microphone to plug into the transmitter, clip into a shirt or jacket lapel, slipping the transmitter into a pocket or clipped to a belt. The broadcast-grade lavalier microphone can also be connected to any camera or device with a 3.5 mm TRS jack mic input that supplies plug-in power, including the RØDElink range of wireless transmitters.

The new Røde design delivers 83 dB typical signal-to-noise ratio (SNR) and is incredibly forgiving when it comes to placement on talent, allowing for greater flexibility in fast-paced situations. It is even compatible with smartphones and other devices with TRRS inputs using the Røde SC4 adapter. The high-quality omnidirectional condenser microphone was completely designed in Røde's state-of-the-art facility in Sydney, Australia and sells for \$79 USD in a package that includes a durable mounting clip for quick and easy rigging, a foam pop filter for reducing harsh plosives and wind noise, a Kevlar reinforced cable, built to withstand even the most demanding usage, and carry pouch.

www.rodemicrophones.com



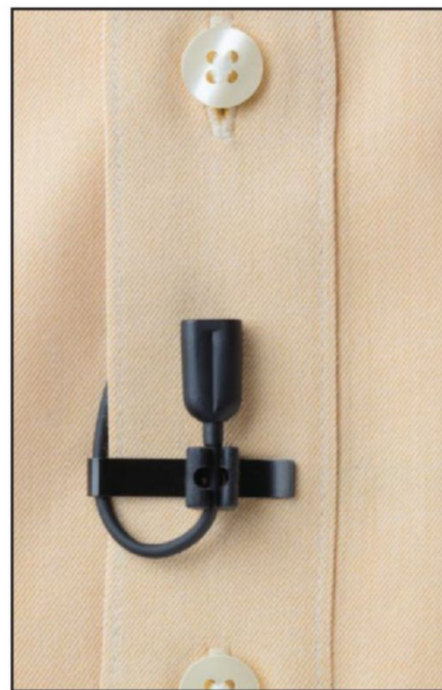
Point Source Audio Patent Awarded to Dual Element Lavalier Microphone

Point Source Audio, an audio equipment manufacturer from Petaluma, CA, specializing in the development of miniature microphones and headsets, has received U.S. Patent No. D850,422 for the CO2-8WL, part of the company's CO2 Confidence microphone series. This design is the world's first fully redundant microphone in a wearable housing, providing an all-in-one solution for critical live sound and broadcast applications that require back-up miking. To ensure live wireless microphone backup in cases of frequency cross-talk, low battery packs, or connector or element damage, audio engineers and sound technicians regularly wire voice talent with two separate microphones and signal paths. This can be a complex, time-consuming workflow that can also lead to inconsistent sound quality and cable clutter, while weighing the wearer down with cumbersome hardware.

With the CO2-8WL, production professionals can mike up talent in half the time and count on stable, consistent audio performance in live, broadcast or streaming scenarios, thanks to the built-in dual element microphone with perfectly matched frequency and sensitivity elements. Both miniature microphone elements are water, makeup, and sweat proof making them ideal for use under hot stage lights and in outdoor productions.

The patent, which covers dual-element microphone design in a wearable housing, is the fourth issued to Point Source Audio in just three years; the company also holds a patent for its CM-i3/CM-i5 modular, lightweight in-ear headsets and two patents for its EMBRACE Earmount microphones, which allows head-worn microphones to be easily concealed on the wearer.

www.point-sourceaudio.com



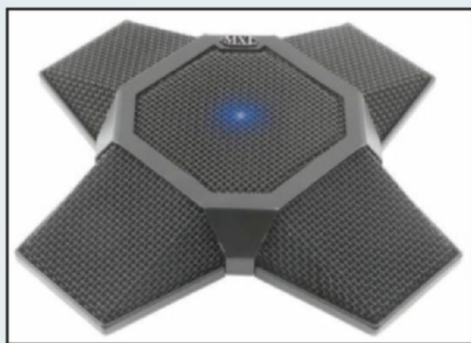
MXL Microphones for Conferencing

MXL Microphones is a California brand that's been successfully riding the UCC market expansion, designing new microphones that work in tight integration with the most popular conference systems and solutions for huddle rooms. Recently, MXL introduced a new advanced "audio tracking" microphone system that can be integrated with select Zoom certified cameras for an improved collaborative conferencing experience. The MXL Zoom Room Speaker Switch System consists of a three-microphone array utilizing MXL's new AC-44 miniature USB microphone (measuring only 2.5" x 3"). The three AC-44s are positioned to deliver a wide pickup arc in a Left, Right, and Center (LRC) arrangement that automatically tracks to wherever each speaker is located. The integrated camera focuses in on each speaker's location based on the audio signals, harnessing Zoom's advanced internal audio processing software to track the current speaker. The three-capsule boundary design makes it ideal for huddle room and conferencing applications. The MXL Zoom Rooms Speaker Switch

System ships with an additional four-port USB extension hub for easy set up and operation.

MXL Microphones also introduced a new universal, table-top web conferencing microphone, the MXL AC-360. The new USB-powered AC-360 works with virtually any video conferencing platform, from Skype to Zoom, GoToMeeting or Lifesize. The AC-360 already features a USB-C interface and builds off the success of its predecessor, the AC-360-Z, which was co-designed with engineers from Zoom Video Communications to be a preferred hardware selection for Zoom Rooms and Zoom Meetings software. The MXL AC-360 uses 12 internal microphone capsules, divided into four quadrants, to ensure 360° of speech intelligibility within a 25' radius, without reliance on any additional equipment. The AC-360 is plug-and-play compatible with any Mac or PC device without needing to download any drivers. Each quadrant of the AC-360 acts as an independent device for optimal performance and fidelity, and each AC-360 microphone ships with a complete installation kit including cables and mounting options.

www.mxlmicro.com



The USB-powered MXL AC-360 microphone is compatible with conferencing platforms including GoToMeeting, Skype, Zoom, Lifesize, and more.



Three MXL AC-44 compact USB microphones can be combined with Zoom's audio tracking software to coordinate with an integrated camera that focuses in on speakers.

AKG CBL Series Boundary Layer Microphones

AKG CBL Series boundary layer microphones are available in multiple form factors and coverage patterns to accommodate critical applications, including conference rooms and meeting spaces. Following the current market trends in those type of applications, the latest AKG CBL series boundary layer microphones supports both analog and digital connections and can be equipped with a four-channel Dante network interface. These microphones are easy to install, easy to use, and feature an ultra-low profile design, ensuring minimal disruption while sitting on a table or desk. The CBL301 is equipped with a triple element cardioid capsule configurable from 270° to 360°, while the CBL201 utilizes a dual-element cardioid capsule fixed at 180°. With a built-in mute switch, users can toggle CBL microphones from talk to mute with the press of a button, while the onboard RGB LED ring indicates the current talk or mute state. With built-in RF interference suppression, CBL microphones prevent interference from mobile phones and laptops.

www.akg.com



Audio-Technica ES954 Quad-Capsule Steerable Microphone Array

Audio-Technica ES954 Hanging Microphone Array is a cost-effective solution for huddle rooms, conference rooms, and other meeting spaces. When controlled by the Audio-Technica ATDM-0604 Digital SmartMixer, this four-capsule microphone array provides 360° coverage through virtual hypercardioid or cardioid outputs that can be steered horizontally and tilted vertically. An high-output design with low self-noise delivers strong, natural-sounding vocal reproduction. The ES954 may be used singly or in multiples to capture every person speaking in a room, with the total number of channels restricted only by the capacity of the mixer or DSP device controlling the system. When using the ATDM-0604 Digital SmartMixer, the intuitive graphic interface enables control of the width and orientation of each virtual polar pattern, which may be steered in 30° increments, with a tilt function accommodating differing ceiling heights or users that are sitting/standing. An integral, logic-controlled 360° red/green LED ring provides clear indication of the array's mute status. The included seismic cable secures the AT8554 Ceiling Mount to a drop ceiling grid.

www.audio-technica.com



Countryman A3 Halo Gooseneck Microphone for Podium Applications

Countryman Associates, a manufacturer of miniature microphones and direct boxes, introduced the A3 Halo Gooseneck Microphone, based on the brand's A3 Podium Microphone. The A3 Halo includes an ultra-bright LED indicator ring and uses a dual-element design that allows for a switch-selectable omnidirectional, cardioid, or hypercardioid polar pattern. This is an updated choice for houses of worship, lecture halls, or legislative chambers. Key among the A3 Halo Gooseneck Microphone's feature set is the Active Plus Vibration Cancellation feature, which dramatically reduces table thumps and microphone handling noise without the need for shock mounting—making it ideal for historic buildings and flexible seating spaces. The in-mic jumper system enables installers to reconfigure the pins of the A3 Halo's XLR5 to match most common halo gooseneck configurations without the need for system rewiring. With a choice of three lengths in a rugged semi-rigid design, the new A3 Halo Gooseneck Microphone also features a screw-on multistage windscreen and rigorous RF protection. The A3 Halo Gooseneck Microphone is available in fixed pattern (hypercardioid, cardioid, and omnidirectional) models, or a selectable pattern model.

www.countryman.com



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A Groundbreaking Approach to Gain Control that Cuts Conference Call Annoyance

Nureva, a conferencing and collaboration-solutions company from Calgary, Alberta, was recently granted three new patents for its Microphone Mist technology, including the ability to focus sound signals in a shared 3D space, and the use of sound source positional information to derive automatic gain control output parameters. *audioXpress* invited Nureva's engineers to explain the thought process that led to these innovations, which give the company substantial market runway.

By

**Erica Yorga
and Rich Ferguson**

Let's face it, conference calls don't get much love. Quite the opposite, often. Why? Could it be that the remote person isn't really paying attention? A *Harvard Business Review* study reveals that 65% of conference callers are doing other work, 55% are making or eating food, and 47% are going to the bathroom.

For onsite people, it might be having to sit still at the table and only speak facing forward as though they were still in grade school (what if you want to work at a whiteboard or just stretch your legs while you're talking?). Constantly being asked to repeat yourself or enunciate more clearly is also extremely tiresome.

But for remote team members, the thing they probably don't like the most about conference calls—and actually find extremely annoying—is how they sound. In any event, this is an audio magazine so that's what we'll concentrate on (and leave pondering people's on-call bathroom habits to the folks at Harvard).

The Effects of Annoying Audio

That feeling of annoyance that many people associate with conference calls is more than just, well, annoying. Bad audio on a conference call (and video calls, too) can make it hard to concentrate and lead to listener fatigue. These effects can inhibit collaboration, clarity, and even morale. Then there is just not hearing people clearly. Misheard words can be costly.

There are also proven physiological effects. *Scientific American* points to studies showing that the negative effects are real. Recent research suggests that "noise-induced" stress caused by annoying sounds can affect the way we think. It can decrease the availability of dopamine in the prefrontal cortex to primary brain functions that ultimately may affect memory and learning.

What's So Annoying about Conference Call Audio?

Why can listening to a person on a typical conference call be annoying, when talking in person

is not (for most people)? Why don't conference calls feel "natural"?

There are a range of factors, like abrupt changes in volume (this happens a lot when a caller is on a cell phone) and audio that can sound tinny or echoey. But one of the most significant factors is background ambient sounds, which are sounds that are heard as background noise as well as any electronic hiss behind a talker's voice on a call. Ambient sounds can interfere with the overall audio experience and the ability to clearly hear a caller.

Here, we are interested in how a given audio conferencing system handles these sounds. This, in turn, becomes the story of automatic gain control (AGC), which is used in most traditional conference call systems, versus position-based gain control (PBGC), which is a new technology found only in Nureva audio conferencing systems. Let's look at how each works and the ramifications for users.

Automatic Gain Control

The purpose of AGC in an audio conferencing system is to keep the audio output level (what the remote person hears) within a targeted minimum-maximum range (set by the system microphone level control). It does this by adding more gain (boosting the signal strength) on quiet talkers and, possibly, reducing the gain on the signal from loud talkers.

This is fine as far as it goes, but there is a problem. The input of an AGC module/circuit treats the audio signal holistically. This means that if the AGC module is required to increase the gain of the audio signal, the whole signal has gain applied to it. This results in the talker and ambient noise being increased equally (see **Figure 1**).

Loud, Quiet, Loud, Quiet: Another AGC Issue

During a conversation with people in the same room (no mics involved), ambient noises are fairly constant. Psychoacoustics shows us how our brains adapt to these background noises. We concentrate on the person talking and filter out the background noises. We can think of this as a "natural listening experience." But with AGC raising the level of the talker and the ambient noise, we get audio moments that are loud (when people are talking and gain is added) followed by moments that are quiet (when no one is talking and no gain is added).

Remote participants might experience the talker staying at the same level (target range) while the ambient noise is heard as increasing or decreasing as the AGC works to maintain a constant talker level for remote participants. This is referred to as "noise pumping" of the AGC module—with the signal-to-

noise ratio (SNR) maintained at the original input signal. This pumping effect can be very noticeable and can be a contributing factor to poor sound quality—which may result in listener fatigue and distraction and a less pleasant experience overall.

Extra processing (including from filters and noise reduction circuits, etc.) is typically required to handle the increase in ambient noise that goes along with the increase in talker volume. This may also result in degradation of sound quality.

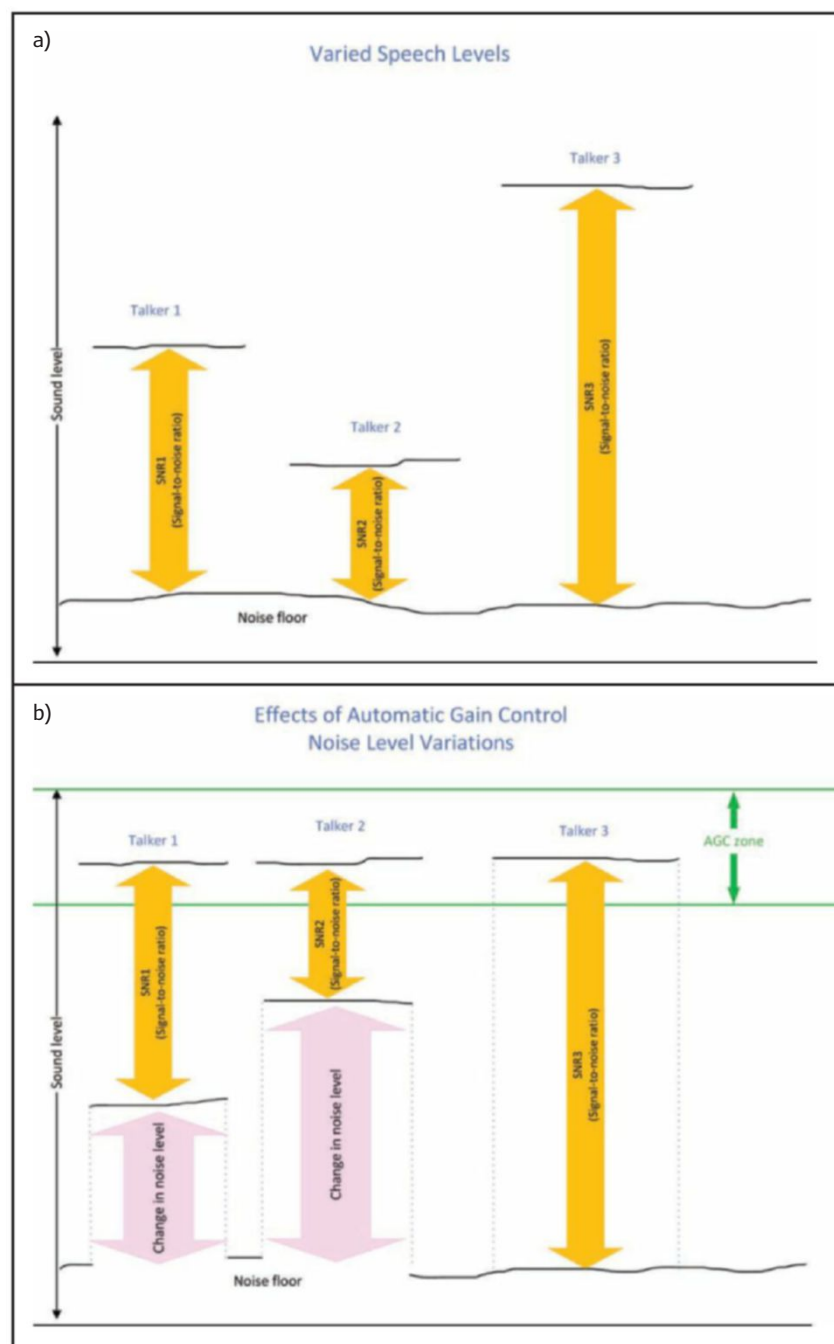


Figure 1: This is the perceived audio level by a remote listener on a system without AGC (a) and the perceived audio level by a remote listener on a system with AGC (b).

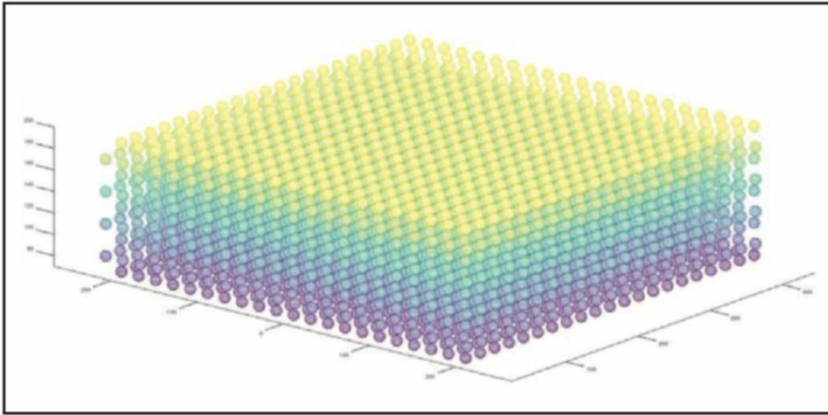


Figure 2: With Microphone Mist technology, each “bubble” is an individual virtual microphone or pickup point. Each of these is a “position” in position-based gain control.

A New Approach to Audio Conferencing

“Can you speak up?” “Can you repeat that?” “Please move closer to the mic.” How often do you hear requests and comments like these on conference calls? Whether they are caused by AGC or not, solving these issues is what motivated Nureva to develop its own audio conferencing systems. And knowing that traditional conference call systems just employed incremental improvements on mature technology, we decided to tackle the problem of bad conference call audio with a completely new approach.

The Microphone Mist Technology Revolution

After two years of blue-sky thinking and intense research and development, our team of scientists and audio engineers came up with a revolutionary concept we call Microphone Mist technology. Unlike traditional beamforming technology that steers pickup to just a few defined areas of a space and leaves dead zones where people can’t be heard, Microphone Mist technology fills a room with

thousands of virtual microphones (see **Figure 2**) so people can be heard no matter where they move in a space or which way they face. A virtual microphone is an individual “pickup point” created when two or more physical microphones work together through proprietary processing algorithms to focus their pickup patterns on a configurable zone.

Thanks to advanced processing power, Microphone Mist technology listens to each virtual microphone (essentially, the “positions” in PBGC) simultaneously and zeros in on the predominant sounds (talkers, typically), optimizing them for clear conversations. You can see the difference between the pickup patterns of traditional beamforming technology and Microphone Mist technology in **Figure 3**.

A New Paradigm: Position-Based Gain Control

This ability to zero in on sounds creates a number of benefits (more on them later in the article), but one of the most significant is enabling PBGC.

Microphone Mist technology focuses on the predominant sounds you want to hear (talkers) while maintaining a relatively constant background/ambient noise experience, optimizing the audio output. This is accomplished through patented algorithms that utilize the measured talker position to determine the correct gain processing strategy (algorithm) that increases the talker level, while maintaining constant ambient noise performance. The result is an increase in SNR that directly translates into a less fatiguing and distracting remote listener experience.

Typical AGC circuits rely on heuristics and expert logic to determine how and when they should apply or decrease gain. An AGC circuit does not know how far away a talker is from the mic, so all signals—good and bad—are treated equally. Since Microphone Mist technology knows the talker’s position, the PBGC module can make specific optimizations to the microphone system and processing of the signal that other systems without positional information cannot (see **Figure 4**). These patented proprietary processes enable Nureva to optimize the talker gain while keeping ambient noise constant.

Natural Listening Experiences

The result is a “natural listening experience” for remote participants. They feel like they are in the room with their teammates—with a natural mix of voices and ambient noises. This makes conference calls far less annoying and fatiguing.

To be honest, we had a bit of a chuckle here at

About the Authors

Erica Yorga is the Principal Intellectual Property manager at Nureva, a Canadian technology company. Her career spans a variety of leadership roles researching, developing and commercializing a broad spectrum of technologies in markets such as audio/visual, oil and gas, telecommunications, military defence and Government regulatory. The challenge of solving difficult problems by constantly learning and enabling teams through the application of unique approaches to technology and process is a passion she brings to every endeavor she undertakes.

With more than 20 years’ experience leading software and firmware development teams, Rich Ferguson has helped build groundbreaking products in storage, interactive collaboration, and communications industries. He has been with Nureva for almost three years and enjoys the challenges involved in developing audio related solutions.

Nureva when a competitor published a blog telling users how they had to change their behaviors—like not moving around the room and not speaking away from the mic array—to get adequate conference call audio. We think asking the user to adapt to technology is exactly backward. It's also far from "natural."

With Nureva audio conferencing, the technology adapts to users, so they are free to move about work spaces, face in any direction, and even talk at the same time. And they can speak in normal conversational voices and still be clearly heard by the remote caller. Definitely more natural.

Advanced Ways to Eliminate Unwanted Sounds

We have also seen others in the industry who ask users to remove or keep ambient noise low—which is somewhere between difficult and impossible. How do you turn off the fan in a system with programmed climate control? This is why Nureva developed technology, called intelligent sound targeting (IST), which enhances our systems' ability to diminish the annoyance of unwanted noises.

While traditional systems try to deal with unwanted sounds using "post-sound" filters that require substantial processing resources to suppress sounds, IST in our Microphone Mist technology takes a "pre-sound" approach. Using adaptive learning algorithms, IST preemptively identifies and learns unwanted persistent sounds and then simply stops picking them up. This is only possible because of Microphone Mist technology's ability to listen to, and delineate, those thousands of virtual microphones.

Are you seeing a pattern here? Another audio conferencing situation that suffers from extraneous noises is a presentation in front of a large group, such as a lecture. Here the main culprit is chatter and other noises from the audience or class. To solve this, Nureva developed an extension called Active Zone Control that can really make a difference for remote students or those viewing a class later through lecture capture.

Active Zone Control enables the presenter or IT to use a simple interface to define the pickup zone in a space. Pickup can be focused on the area where the presenter will move at the front of the room while pickup from the virtual mics in the rest of the space is sharply diminished. Bonus: the presenter gets isolated voice pickup without the hassle of a lapel mic or headset.

Now Hear the Difference

We began this article by noting the potential for real negative effects from bad conference call

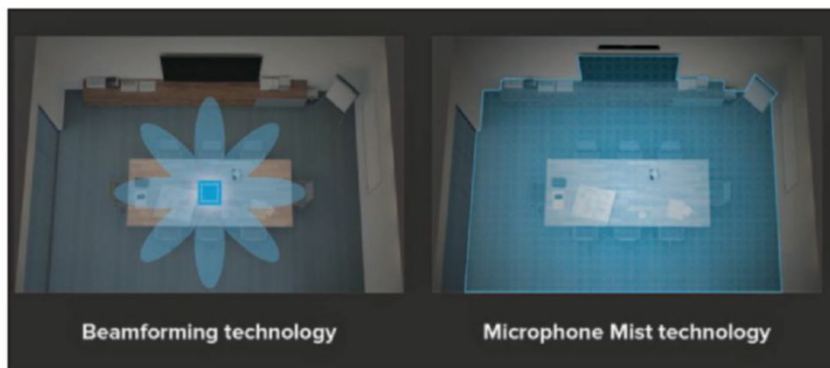


Figure 3: Note the difference in pickup coverage between a system using beamforming technology (partial pickup, with dead zones) and one with Microphone Mist technology (true full-room pickup).

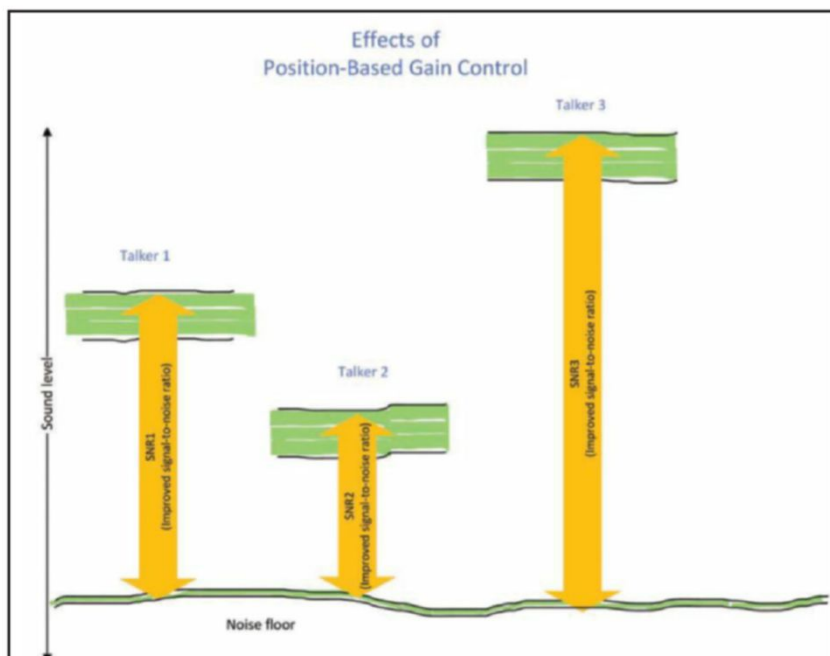


Figure 4: Position-based gain control (PBGC) raises the levels of talkers so they can be heard more clearly while only slightly raising the noise floor. And the noise floor remains more consistent—without sharp drops or steep inclines in the audio level that can cause annoyance and fatigue.

audio, especially from annoying ambient sounds. As we've shown, new technology has the potential to mitigate these effects and help team members on both ends of a conference call have a more natural listening (and speaking) experience.

You can visit [Nureva.com](https://nureva.com) to learn more about Microphone Mist technology and our line of audio conferencing systems. But best of all, we have such confidence in the performance of our technology, we offer live online demos (we are one of the few audio conferencing solution providers to do this) so you can hear the difference a natural listening experience makes. 

Positioning Choir Microphones



Among the most challenging aspects of sound-system design and installation for traditional auditoriums and worship centers is choosing and positioning choir microphones. Looking at existing installations in many venues, one is forced to assume that the installers were working from the bliss of ignorance, or at best using the time-dishonored “By Guess and By Gosh” (BGABG) method. The good news is, there is a right way to choose and use choir mics, and this article will steer you in that direction.

By
Richard Honeycutt
(United States)

Obviously, the source of the sound you want the choir mics to pick up is human voices. So it's important to know something about the acoustics of the human vocal system, especially the directional characteristics. Of course, the lower frequencies are pretty much omnidirectional, radiating equally to the front, back, top, and sides of the body. But most of the vocal quality relates to the harmonics, which are higher-frequency components. **Figure 1** shows a simplified diagram of the directional characteristics of a human singer or speaker. Standing 20, 50, or 100 singers side by side will not change the directional pattern. We'll come back to this point later.

The Acoustic Environment

In a perfect world, not only would the choir be made up of professionals who recognize that the sound tech is their friend, but the choir loft would be sound-friendly: the ceiling would be reflective, so that the choir members can hear each other; the ceiling would be low enough that the reflections would reinforce the sound of the choir rather than

becoming bothersome echoes. The wall behind the choir would also be reflective in order to help project the natural sound of the choir toward the listeners. The floor of the platform in front of the choir would be hard wood in order to reflect the vocal sound (which, as you remember, is aimed downward) toward the listeners. The floor directly under the choir members' feet would be carpeted to reduce foot-shuffling noises.

For now, however, we must work in this world, where conditions are not always so perfect. So what good does it do for us to know what perfection would look like? Well, one thing we can do is to see whether there are any possible acoustical improvements that we can influence. If heavy drapes are used beside or behind the choir, we can suggest that they be removed. If there is acoustical ceiling above the choir, we can suggest that it be replaced with something less absorptive.

The goal here, as always, is good sound. And anything we do to improve the natural sound of the choir will improve the sound we get through our mics.

Top photo: The 4097 CORE supercardioid choir microphone from DPA Microphones was designed specifically to capture dynamic choir sound, with both wireless or wired configurations.

The Mics: Requirements

Now we're ready to talk about the mics. First, what do we want them to do? Basically, there are six requirements:

1. The mics should pick up all the choir members evenly.
2. The mics should reject any sounds not coming from the choir (or instrumental accompaniment).
3. The mics should have accurate frequency response.
4. The mics should have very low self-noise.
5. The mics should be completely invisible.
6. If an orchestra or pipe organ is used, the mics should pick up the accompaniment and the choir in proper balance.

Of these requirements, #1 is the most important to the choir director; #2 matters most to the sound tech; #3 is most noticeable to the listeners; #4 will be noticed most by the engineer at the radio or TV station if the program is broadcast; #5 is by far the most crucial to the Sanctuary Decoration and Nitpicking (SDN) Committee if the venue is a church; and #6 matters to everybody. As it turns out, some of these requirements conflict with each other. But recall our earlier observation about working in this imperfect world.

Requirements #1 and #2 relate to a characteristic of the microphones themselves and to their placement. That mic characteristic is directionality. **Figure 2** shows the most common mic directional patterns, along with the "acceptance angle" for each.

Most often, meeting requirements #1 and #2 involves using cardioid or hypercardioid mics. The mics should have a flat frequency response from at least 80 Hz to 10,000 Hz, free from presence peaks (and "absence dips"?).

So the first choice to be made is:

- One of the excellent mics designed specifically for choir use by Countryman, EV, Shure, Audio-Technica, DPA, Røde, or other manufacturers is employed. Designed to be inconspicuous, these come in two basic shapes. One is a small cylinder and utilizes a specially-shaped wire fixture to point the mic at the choir. The other is a tiny rectangle which is active on one face. (Or both faces, in the case of a bidirectional model.)
- Another option is to use a small cylindrical mic as illustrated in **Photo 1** and put up with greater visibility.

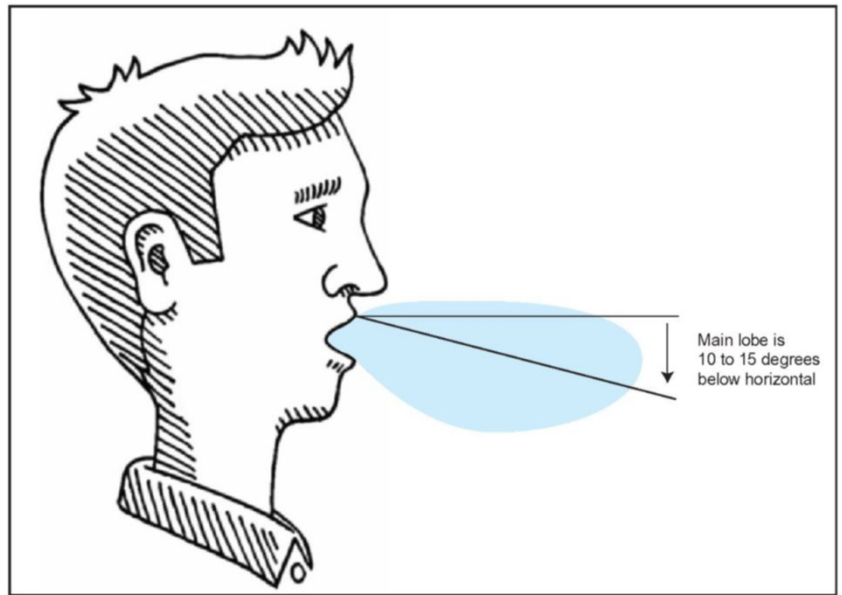


Figure 1: The directional pattern of the human voice is typically out and downward.

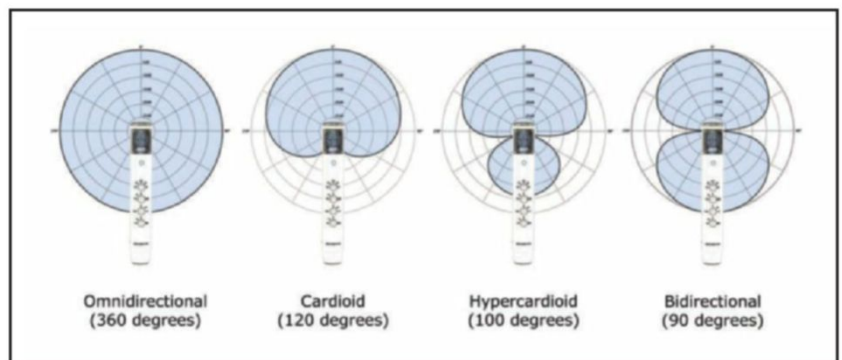


Figure 2: Here are a few common microphone directional patterns.



Photo 1: The Audio Technica AT2021 small cylinder microphone is suitable for choir and orchestra pickup.

Distance Ratio	Imbalance
2:1	6 dB
1.78:1	5 dB
1.58:1	4 dB
1.41:1	3 dB
1.26:1	2 dB
1.12:1	1 dB

Table 1: Here we have the distance variation versus the decibel imbalance.

Any of these choir mics can be used with small foam windscreens, but these are seldom needed unless the mic must be hung in the stream of an HVAC vent.

Positioning the Mics

We've already begun talking about suspending the mics. The alert reader will have noticed that a suspended mic is not in the ideal position to pick up voices that have the directional pattern shown in Figure 1. Unfortunately, the other options are usually worse. Let's examine them.

Footlight mics (e.g., the Countryman DBM or the various pressure-zone or boundary mics) are a bit too low for optimum pickup, but are better positioned than suspended mics. However, footlight mics are liable to be stepped on or kicked, or at the very least they will pick up footstep noise. And you can never be certain that the SDN Committee won't decide to place a huge potted plant just in front of the mic.

Stand mics can be positioned at the exactly correct location, but are subject to the same problems as footlight mics, besides being visually distracting. In addition, stand mics experience comb filtering when used with distant sources (over 1' or so) because of interference between the direct sound and the sound reflected back up

from the floor. This imparts a hollow sound quality that is really unpleasant.

So we are usually stuck with suspending the mics, even though we know that puts them off-pattern for the singers' voices. But where exactly do we put them? Here are five potential considerations:

1. Give even pickup of all the choir members
2. Avoid overlap of various mics' pickup patterns
3. Get the mics as close to the choir as possible for the best presence and highest gain before feedback.
4. Provide adequate clearance above the floor for the mics' physical safety.
5. Offer balanced pickup of choir versus accompaniment.

The direct sound from a choir member will decrease at a rate of 6 dB with doubling of distance outdoors or in a very absorptive room. If the choir loft is properly designed acoustically, with reflecting surfaces close to the choir, the decrease will be less than 6 dB. But to be conservative, we'll use the 6 dB figure. This means, for example, that if Albert is twice as far from the mic as Alice is, Alice's voice will be 6 dB louder in the mix. Not knowing Albert and Alice, we cannot state whether this is a good thing or not, but in general, we'd like for each voice to come through equally. **Table 1** shows the ratio of nearest-to-farthest distances from choir member to mic that correspond to various decibel values of imbalance.

Thus if we insist upon having the various choir members' voices picked up evenly (within 2 dB of each other), and the one nearest the mic is located 10' away, the one farthest from the mic must be no more than $1.26 \times 10 = 12.6'$ from the mic. As you will appreciate, this can be challenging to arrange. In fact, it turns out that about a 6 dB variation is pretty much the best for which we can design.

When two mics are positioned so that they both pick up the same sound source, there will be interference between the mics' output signals when they are mixed. At any given frequency, the difference in path length from the source to the two mics will translate into a phase difference in their outputs. This phase difference will cause the signals at some frequencies to add, while at other frequencies, they cancel. The resulting frequency response plot is a series of alternating peaks and dips that looks like a comb. Hence this problem is called comb filtering (see **Figure 3**).

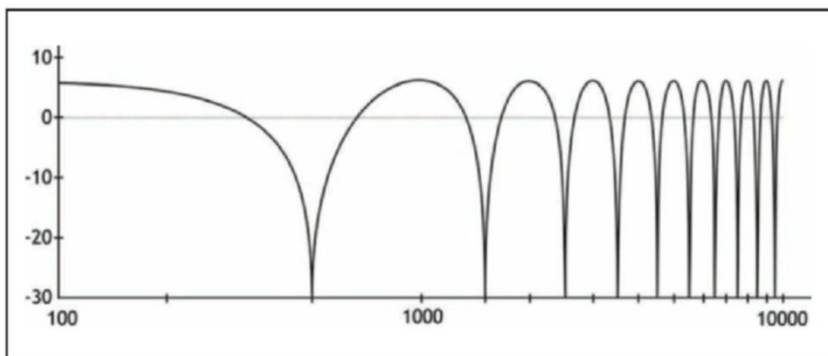


Figure 3: Comb filtering occurs when the same sound is picked up by two microphones.

Comb filtering just sounds weird. It also decreases gain-before-feedback.

Some 22 years ago, my company installed the sound system for a new church in a nearby city. In this particular church, the chain of command was the pastor. He wanted very even choir pickup, but knew nothing about mics, so he absolutely insisted that we use five mics to cover a 35' wide choir loft. The result was as bad as I told him it would be. I did succeed in getting the operator to understand that he should not use all the mics at the same time. Thus, we kept the comb filtering somewhat under control while still giving the pastor what he insisted upon.

The general guideline for avoiding comb filtering is that the distance from a given source to the nearest mic should be less than one-third the distance from that source to any other mic. This is often called the 3:1 rule. In choir miking, this is usually an impossible condition to achieve.

The best we can do is to use the fewest mics we can get by with, and design for a small underlap in coverage between mics. In other words, we

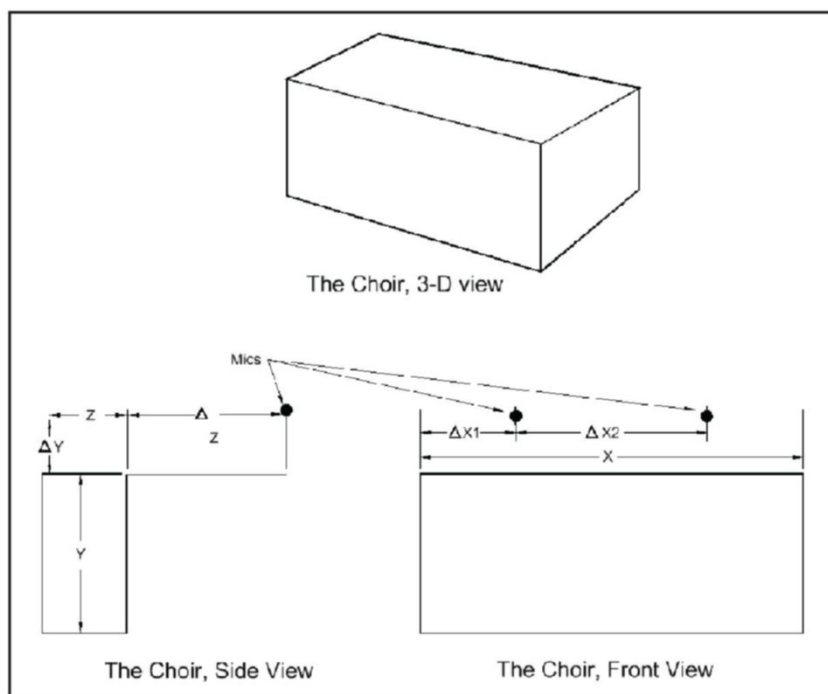


Figure 4: This diagram shows proper choir microphone placement.

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Stereo Pickup Systems	Microphone Types	Microphone Positions	
X-Y	2-Cardioid	Axes of maximum response at 135° spacing: coincident	
ORTF (French Broadcasting Organization)	2-Cardioid	Axes of maximum response at 110° spacing: near-coincident (7")	
NOS (Dutch Broadcasting Foundation)	2-Cardioid	Axes of maximum response at 90° spacing: near-coincident (12")	
MS (Mid-Side)	1-Cardioid 1-Bidirectional	Cardioid forward-pointed; bidirectional side-pointed; spacing: coincident	
Spaced	2-Cardioid or 2-Omnidirectional	Angle as desired; spacing: 3' to 10'	

Table 2: Coincident microphone pairs help address sound and spatial issues.



Photo 2: These portable digital recorders incorporate coincident microphones.

separate the two mics a bit “too far” for even coverage. The result is that the singers whose voices are subject to comb filtering are not picked up as strongly as the others, so that the sound degradation caused by the comb filtering is reduced. This approach also reduces the physical width of the comb-filtered region of the choir. (If we tell the choir director what we’re doing, (s)he can often use this underlap to advantage by placing the self-appointed divas in those locations.) Naturally, the fewer mics we use, the farther they have to be from the choir in order to get even pickup. So again we have a conflict in design goals.

Next, just what is adequate clearance for the choir mics’ physical safety? Some church members may be close to 7’ tall. Yeah, only very few, but you don’t want anyone’s head bumping the mic. Also, some wedding directors in churches like to place burning candles directly under the choir mics (I suspect they’re secretly in the pay of the SDN Committee), so some height is needed to avoid cooking the mics.

Positioning Spaced Suspended Mics

First, don’t hang the mics directly above the choir, as do some installers I’ve seen. While you can get pretty even pickup from there, the sound is awful. Try holding a mic above a singer’s head and see how good a sound you can’t get!

Figure 4 illustrates the placement of a choir mic, with the important distances labeled. It turns out that dimension Z, the depth of the choir (front row to back row) is very important. In order to pick up each member, with a variation under 6 dB, the mic should be about 8’ from the floor, and the horizontal distance ΔZ from the front row to the mic should be about 1.4 times Z.

If more than one mic is used, the distance between the mics (ΔX_2) should be 2.9 times Z for even pickup of the front row. This will give a bit of underlap for the rows behind the front, as we discussed earlier. For more underlap, you can space the mics a bit farther apart, but the $2.9 \times Z$ works well in most cases. If the floor of the choir loft is angled, you can raise the mic to 9’ or 10’ feet and move it a foot or two closer to the choir. The mic should be no farther than $1.2 \times Z$ from the end of the front row (dimension ΔX_1). The reason that the distance between mics should be over twice the distance from a mic to the end of the row is that the choir members between the mics are picked up by both mics.

This arrangement allows each mic to cover a distance of about $2.65 \times Z$ of the choir width.

Resources

E. Tarr, “4 Common Stereo Microphone Techniques,” Pro Audio Files, April 2015, <https://theaudiofiles.com/4-stereo-miking-techniques>

For a choir arranged in four rows, the depth of the choir is about 10', so each mic would cover about 26.5' of choir. A hypercardioid mic will provide just enough coverage, but if the small rear lobe of a hypercardioid is a problem, you can use a cardioid mic. An omnidirectional mic is OK for recording, but will have less presence, and will provide less gain before feedback in a sound-reinforcement application.

Now you may be thinking that hanging the mic 14' horizontally from the front of the choir puts it a bit far away. That would be true if the choir has only one or two rows. But as the choir depth increases, the distance from the mic to the singer in the back corner becomes proportionately greater than the distance from the mic to the nearest singer, and that back-corner singer begins to disappear from the mix. So you see, the distance from the mic to the front row really has much less effect on the balance among singers in the front row than it does on the balance of front row to back corner.

You can use more mics to improve overall side-to-side balance, but remember that you will be introducing comb filtering in the area of the choir where the mics' pickup ranges overlap. Usually any improvement in evenness of coverage is more than offset by the results of the comb filtering.

For recording only, you can place the mic farther from the front row of the choir and you will get less variation. A maximum variation of 2 dB requires the mic to be about $4.2 \times Z$ in front of the choir, which is pretty far away! But in an acoustically dead room, sometimes you can get away with it.

Coincident Mics

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

In order to address this and other issues, various coincident and near-coincident mic techniques have been developed, as illustrated in **Table 2** (see Resources). Since the mic diaphragms are located very near to each other, comb filtering is not a problem. The ORTF or X-Y coincident pair is incorporated in some portable digital recorders, as shown in **Photo 2**. The weakness of coincident pairs is that their stereo separation is limited.

By using these guidelines, you can be sure that the choir pickup will be as even and natural as possible, you will have a sense of a job well done, and your company's reputation will be enhanced. Maybe even the chairperson of the SDN committee will contact you for a system rental for the Spring Flower Show!

In a future Sound Control article, we will discuss miking choirs that perform with an orchestra or pipe organ. 



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Interfacing Measurement Mics to Delta-Sigma ADCs

Nuances Applying FDAs

Our audio expert details some of the subtle issues associated with the latest fully differential amplifiers (FDAs). Subsequent Audio Scope articles will detail active filtering options and phase margin improvements.

By
Michael Steffes

(United States)

Measurement microphones provide a 2 Hz to 50 kHz audio span stretching the limits of legacy amplifier solutions to deliver a low power interface to 24-bit ADCs with the requisite <130 dBc harmonics at the ADC inputs. Newer solutions, using the latest fully differential amplifiers (FDAs), can easily fill this gap using very low power—but they bring with them some new and subtle issues that can trip up even the most experienced designers. Those newer nuances will be addressed in future articles.

Figure 1 shows an example single-ended input to differential output with typical filter elements using best-fit standard value RCs driving the PCM4202^[1] dual-channel 24-bit audio ADC. This is an AC coupled, third-order Butterworth, low-pass

design using a second-order multiple feedback (MFB) active filter with a first-order RC at the ADC inputs. This nominal gain of 1.7 V/V design has targeted $\approx 1\text{ k}\Omega$ input resistor with resistors chosen that give slightly higher gain using E96 values. The 65 kHz $F_{-3\text{dB}}$ target yields a slight 0.84 dB roll-off at 50 kHz but greater than 100 dB attenuation by 6 Mhz. These required pole locations are coming from the Burr-Brown Filterpro design software.

The Fully Differential Amplifier

The 7 mW, very low distortion, THS4551 FDA^[2] brings with it a number of new design considerations. But first, where in the world did the name "Fully Differential Amplifier" come from? The earliest

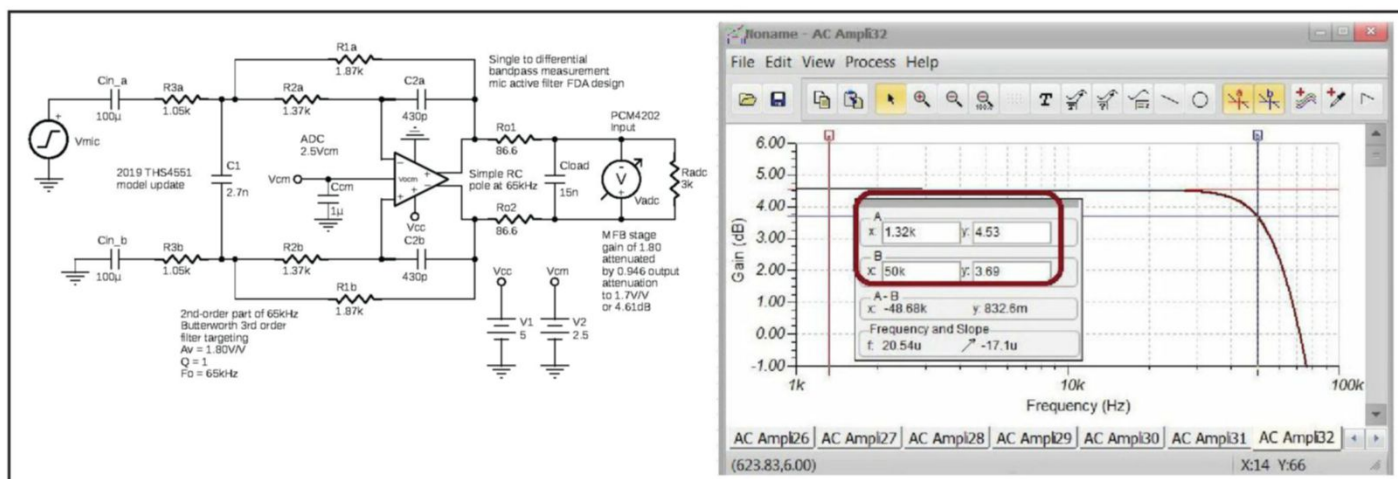
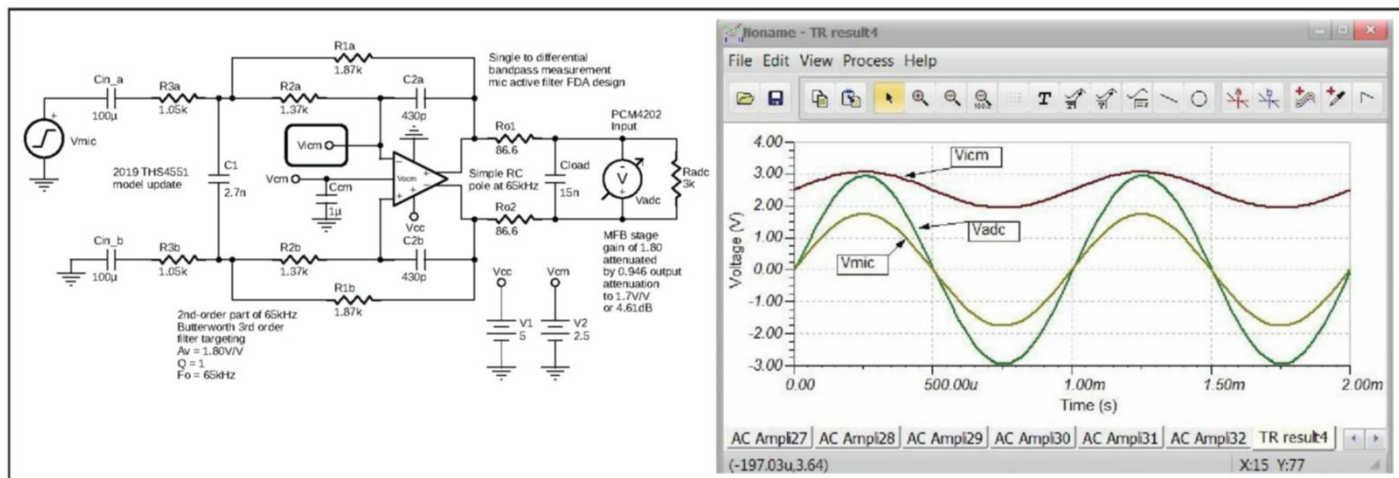


Figure 1: In this example, a single-ended input to differential output with typical filter elements is shown using best-fit standard value RCs driving the PCM4202 dual-channel 24-bit audio ADC. Notice the 2 Hz to 65 kHz third-order Butterworth single to differential FDA-based active filter solution.



introductions for this fundamentally new type of amplifier^[3] used the simple “differential amplifier” nomenclature. By the late 1990s, there were already a wide range of “differential” amplifier application solutions. The earliest Texas Instruments (TI) introductions, including the THS4131 (same die as the audio OPA1632^[4]), coined the marketing term “Fully Differential Amplifier” to distinguish it from dual op-amp solutions. The various suppliers^[5] now use

this “Fully Differential Amplifier” name to distinguish this fundamentally new high-speed amplifier topology from other types of differential output solutions.

The FDA (as it is used now) is two amplifiers working together to offer a single to differential conversion stage with an independently set output common mode voltage. This solution was driven by the rapid migration through the mid-1990s to differential input ADC’s. All current ADCs require a controlled input

Figure 2: In this example, the 1 kHz full-scale input/output voltage swings with the input common mode voltage swing.

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common mode (CM) voltage for proper operation with differential inputs swinging around that CM voltage.

The FDA is a differential I/O classic voltage feedback amplifier (VFA), or current feedback amplifier (CFA), with an added CM control amplifier. That second amplifier senses the output average (common mode) voltage and compares that to an internally default mid-supply, or externally applied, common mode reference voltage and adds an error signal internally at the inputs to servo the output common mode voltage to that desired target CM. Internally, that CM control loop is very fast (on the order of the closed loop differential signal path), but is often bandwidth limited at the V_{OCM} control input pin to a much lower bandwidth for noise reasons. This new topology brings a lot of advantages to driving differential ADC driver—along with some new design issues.

The FDA is an extremely flexible topological solution. Often used as a single to differential stage, it can also operate purely differential input/output. The output common mode DC voltage can be left to default to mid-supply in most FDAs—or driven by an external reference voltage like the PCM4202 V_{CM} reference output of 2.5 V shown in Figure 1. This driven input V_{OCM} is preferred in this application to reduce the $>600 \text{ nV}/\sqrt{\text{Hz}}$ V_{CM} noise (at $f < 100 \text{ Hz}$) for the floated input V_{OCM} in the THS4551. Output CM noise contributes to the differential output noise through feedback ratio mismatch.

The DC operating voltages around the I/O pins must first be set correctly. This audio application is unique in that the input can be AC coupled (those 100 μF tantalum input capacitors shown in Figure 1 setting a 1.6 Hz high-pass pole), while the outputs are best left DC coupled to provide the ADC common

mode voltage. Here, the V_{OCM} control input is driven by the PCM4202 reference output voltage at 2.5 V. Since the paths from the output CM controlled DC voltages back through the feedback networks are AC coupled, that CM loop DC biases the two FDA input pins at the same output CM voltage. This mid-scale input DC bias is normally safe, but a small input common mode voltage swing will be present in this application. That swing needs to be considered for input clipping constraints.

Most audio ADCs appear to be looking for a $2 V_{RMS}$ input, or a $5.6 V_{PP}$ for a single-tone full-scale differential input. The gain of 1.7 V/V design shown in Figure 1 then requires a $\pm 1.65 V_{PP}$ single-tone input to exercise full scale. Adding an input pin common mode voltage sense meter gives the 1 kHz full-scale sinusoidal simulation shown in **Figure 2**.

The I/O swings look correct with the FDA to ADC connections swapped to give an in-phase output for this normally inverting MFB filter. The FDA input pins (V_{ICM}) are sitting at 2.5 VDC with a $\pm 0.5 \text{ V}$ CM swing on each input pin. This is an artifact of the single to differential configuration where that V_{ICM} input swing generates the inverted side output swing from the lower half of the circuit shown in Figure 2. That maximum input pin voltage of 3 V is well in range considering the THS4551's 1.3 V maximum headroom to the +5 V supply used here. Most recent FDAs suitable for low power audio are negative rail input with rail-to-rail outputs.

Since the V_{ICM} voltage is moving in phase with the input signal, the single-ended input impedance is actually an active impedance (Z_A) scaled up from the physical input R_G (or R_{3A} here) by the CM feedback loop. Equation 1 gives that impedance where $\alpha = \text{gain} = R_F/R_G$ (R_1/R_3 shown in Figure 1). Using $\alpha = 1.78$ from Figure 1, gives an apparent input impedance of $1.05 \text{ k}\Omega \times 1.47 = 1.54 \text{ k}\Omega$. This will be the apparent load to the measurement mic while the value of R_3 is used for the filter response analysis. This effect increases the apparent input impedance to the mic over just the physical R_3 , improving the harmonic distortion performance for the measurement mic.

$$Z_A = \frac{2R_G(1+\alpha)}{(2+\alpha)} \quad [1]$$

Continuing next month with the example design shown in Figure 1, we will explore what error sources need to be considered along with filter response options.

Author's Acknowledgment: Thanks to Taoufiq Bellamine, (Director of Engineering, Earthworks Audio), for his many useful insights driving these audio delta sigma ADCs.

References

- [1] "High Performance 24-bit, 216 kHz Sampling Stereo Audio ADC," Texas Instruments PCM4202 Datasheet, revised September 2004, <https://www.ti.com/lit/ds/symlink/pcm4202.pdf>
- [2] "Low Noise, Precision, 150 MHz, Fully Differential Amplifier," Texas Instruments THS4551 Datasheet, revised July 2017, <http://www.ti.com/lit/ds/symlink/th4551.pdf>
- [3] "Low Cost, High Speed Differential Amplifier," Analog Devices Datasheet, Revision I, <https://www.analog.com/media/en/technical-documentation/data-sheets/AD8132.pdf>
- [4] "High-Performance, Fully-Differential Audio Operational Amplifier," Texas Instruments OPA1632 Datasheet, revised October 2015, <http://www.ti.com/lit/ds/symlink/opa1632.pdf>
- [5] "Ultralow Power, Low Distortion, Fully Differential ADC Drivers," Analog Devices ADA4940 Datasheet, Revision E, https://www.analog.com/media/en/technical-documentation/data-sheets/ada4940-1_4940-2.pdf

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CELESTION

Vanatoo Transparent One Encore

Stuart Yaniger discovered Vanatoo's Transparent Zero a few years ago and still uses those speakers in his home today. Now, he is putting Vanatoo's latest offering, the Transparent One Encore (T1E), to the test to see how it stacks up.



Photo 1: The Vanatoo Transparent One Encore is a very conventional looking small two-way loudspeaker, but there's a lot of technology packed inside.

By
Stuart Yaniger

(United States)

Photography by Cynthia Wenslow

A few years ago, at a very crowded and noisy AXPONA, a couple of things stood out. One of them was the Vanatoo-powered speakers, the Transparent One (T1) and the Transparent Zero (T0), which were small, inexpensive, versatile, and clearly had some serious engineering behind them. I was particularly impressed with the T0, which disappeared in the room and had a very natural sound on acoustic recordings—at least compared to a lot of the gigantic speakers on (loud) exhibition. The T1 was good, but didn't grab my attention the way the T0 did—perhaps it was the next-generation engineering in the T0s that did it?

The T1 was Vanatoo's initial product, and although it was a powered speaker, it still used a conventional passive crossover to the drivers, driven by a single amplifier per channel. The T0 went a step forward and contained separate tweeter and woofer amplifiers in each channel, the crossover being done in the digital domain. This improvement carries several significant advantages: crossover

slopes can be much steeper to reduce distortion, delay and phase correction between drivers can result in improved polar patterns, driver/cabinet frequency response errors can be easily corrected digitally, and the crossovers can be modified and updated without changing components. The results of this engineering approach can be seen in my review of the T0 (see Resources), and those speakers are still in daily use here.

I wondered what would happen if Vanatoo applied this approach to an improved version of the T1, and apparently the Vanatoo design team of Gary Geschellen and Rick Kern were already headed down that path. The result is the Transparent One Encore (T1E), which incorporates all these improvements and more. Retail price for the T1Es is \$599 per pair.

Inside the T1E

Compared to the rather oddly shaped T0, the T1E is a much simpler and conventional rectangular

box measuring 10" (25 cm) tall, 6.5" (17 cm) wide, and 8" (20 cm) deep (see **Photo 1**). My review pair were finished in an unobtrusive satin black, but for an upcharge, you can get a cherry wood veneer. The front panel sports a woofer-midrange with a 5.25" (13 cm) diameter concave aluminum cone and a 1" (2.5 cm) dome tweeter. The tweeter has a grill protecting it, which may also have a function controlling dispersion. Crossover between the woofer and the tweeter is at 2 kHz, a rather low frequency for a 1" dome tweeter, but potentially useful for reducing off-axis response gullies as the woofer starts getting more directional.

The woofer has an interesting design, incorporating the XBL technology from Acoustic Development International (see Resources). The XBL technology goes beyond conventional overhung vs. underhung motor assemblies and uses a split dual-magnetic structure and a short coil. As the coil leaves one of the magnetic fields, it enters the other, which flattens out both the BL and inductance vs. displacement curves, linearizing the driver. The convex dust cap dome used for the T1's woofer has been replaced by a smooth concave dust cap; Vanatoo claims that this change is purely cosmetic and has no acoustic effects in the woofer's passband.

The woofer is a very heavy and solid unit, with the area behind the cone damped by the acoustic resistance of a perforated metal basket (see **Photo 2**). The large magnet suggests that the driver has high electrical damping as well.

On the back panel (see **Photo 3**), there's a passive radiator, which takes over from the woofer below the box tuning frequency, extending the bass response. My own prejudices are in favor of passive radiators rather than ports, since the former do not suffer from moving air noises (chuffing), pipe resonances, or reflections from the enclosure's back wall. The T1E's passive radiator has a higher mass than the one used for the original T1, which is used to lower the box tuning frequency from 48 Hz in the T1s to 40 Hz in the T1Es. This gives the T1E slightly better bass extension than its predecessor. The bass roll-off is also shaped to give higher output and lower distortion by increasing the roll-off slope below the tuning frequency from the usual (for a reflex or passive radiator system) 24 dB/octave to 48 dB/octave.

Likewise, the tweeter is new—an aluminum dome replaces the older version's silk dome. Vanatoo tells me that the older driver performed well, but there were some inconsistencies from unit to unit. This isn't surprising given the variability of natural materials like silk and the inherent variability of the coating/doping process. Aluminum domes are



Photo 2: The T1E woofer is solidly constructed and has significant acoustic damping.

easier to make consistently, and the performance data, as we shall see, show no signs of the bugaboo of metal domes, ringing at the top of the audio passband. These are remarkably high-quality drivers for a loudspeaker in this price range.

As with the T0, the T1E houses all of the electronics in one of the speakers, with a cable connection between the "live" speaker and the other channel. The



Photo 3: The back panel sports a passive radiator, an array of input options, and basic tone and volume controls.

Photo 4: The T1E's remote control is relatively compact, convenient, and intuitive to use.



amplification, which is entirely Class D for efficiency and performance, has crossover, equalization, and delay built into it, and is scaled for the efficiency of the respective drivers. The woofer amplifiers put out 100 W maximum, with the tweeters being driven with 12 W. The digital signal processing (DSP) in the electronics also includes limiters and compression, which kick in at high power to prevent driver damage when the speakers are being pushed hard. These limiters can be disabled by the user, but that can result in the overdriven speakers distorting rather than smoothly compressing. In my case, I never had them loud enough to have the limiting kick in, but if you're fond of concert rock at ear-splitting levels, the choice between a rather smooth compression vs. gross distortion on peaks and bass-heavy material is yours to make. I would be cautious about turning this function off purely from a reliability standpoint; burnt out drivers are an expensive repair.

The T0 used an Ethernet cable to connect the two speakers, whereas the T1E uses a hefty-feeling cable with screw-on connectors. The speakers are supplied with a 6 m long cable, which should suffice for the majority of installations, but Vanatoo can also supply a 2 m or 10 m long cable, if needed. Using a slide switch on the rear panel of the "active" speaker, the user can select whether the electronics-containing speaker is used for the left or the right channel.

The back panel also has a variety of input options, including standard line-level single-ended analog (3.5 mm stereo plug), USB, coax, and optical digital inputs, as well as Bluetooth capability using aptX for higher resolution. A nice touch is that the inputs can either be manually selected or automatically detected. The digital input will accept up to 24-bit and 96 kHz sample rate. All internal processing is done at a 48 kHz sample rate, which exceeds CD standards and is good enough for audible transparency.

There's an enormous flexibility in setup. Besides the left-right swap I mentioned earlier, the speakers have two selectable equalizations for different placements. The "Flat" setting is suitable for stand placement away from room boundaries (the way I used the speakers). "Shelved" will reduce bass output for placement against walls, on shelves, or on a desktop, where the boundary creates bass reinforcement in order to yield an overall flat bass.

Speaking of bass, the T1Es have an output for active subwoofers if extension below the T1Es' -3 dB point of 48 Hz is necessary. When a subwoofer is connected to that output, the T1E automatically sets a high-pass crossover point. In Flat mode,

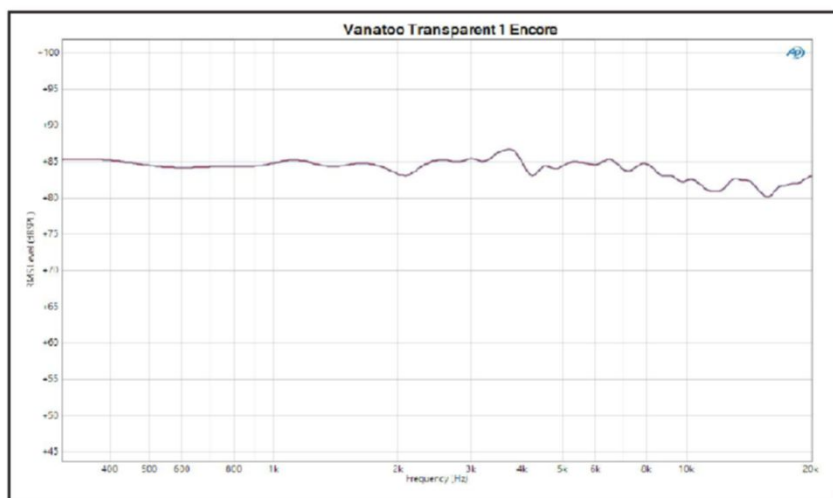


Figure 1: The on-axis quasi-anechoic frequency response of the T1Es is reasonably flat except for a droop in the top octave.

that point is 80 Hz; in Shelved mode, it changes to 125 Hz.

Rounding out the rear panel, the active speaker has an integral line cord, which I personally find somewhat less convenient than a plug for an IEC cord, but this is a minor annoyance. There's also a USB port marked "service," which can provide 5 VDC at 2 A to any external accessory. Additionally, as firmware updates become available, they can be loaded into the speakers' DSP via a thumb drive plugged into that port.

One of the minor complaints I had with the T0 model was the method to change parameters. Each parameter change (e.g., bass, treble, EQ, limiters, left-right swapping, LED brightness, stereo/mono) required a rather elaborate procedure of powering down, waiting a minute or two, setting the volume control to a specific point, setting a three-way switch to a specific setting, then powering up while depressing the Bluetooth pairing button. The three-way switch was needed to reconfigure the single control to bass, treble, or volume. This has been improved somewhat in the T1E, which now has separate volume, treble, and bass controls, as well as a physical switch to determine whether the active

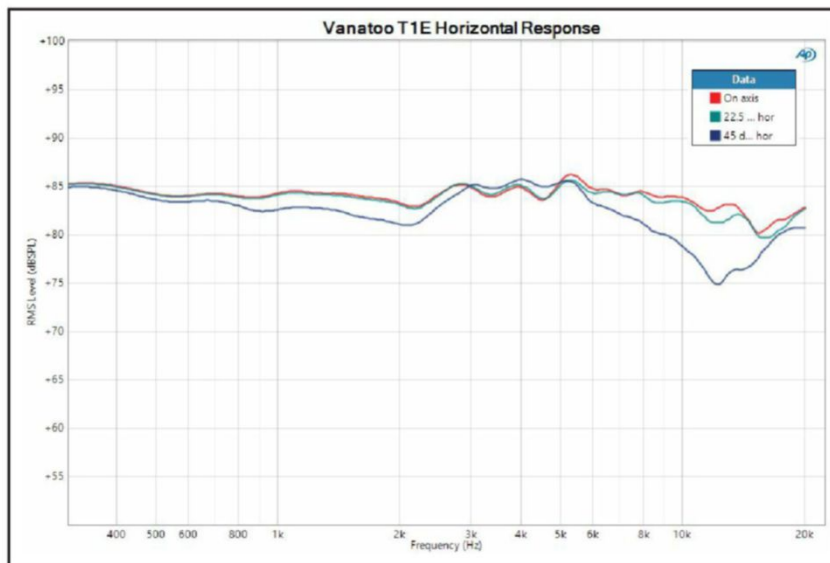


Figure 2: The horizontal dispersion is excellent at small angles, but shows some significant peaks and dips at larger angles.

About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as a technical director for a large industrial company. 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.

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speaker is on the left or right, though the elaborate procedure for other options remains.

Like Vanatoo's earlier speaker models, the T1E's volume, source, and bass/treble can also be adjusted with an included remote control (see **Photo 4**). The controls are logical and intuitive—a nice feature is that the functions within the lower block outline are only active after the user first pushes the "Enable" button. The user then has a 15-second window to make tone adjustments or pair/unpair Bluetooth sources. This prevents clumsy fingers (who, me?) from accidentally changing parameters causing the user to wonder, "What went wrong?" The tone

controls have a "mid" option, which returns them to a flat setting. Pressing the + and – buttons increases and decreases the bass or treble in 1 dB increments.

If I were to make a suggestion to the Vanatoo folks, it would be to reduce remote clutter (we have three of them already in our living room TV/audio setup) in favor of a smartphone app. This could also reduce the clumsy parameter-change method described above. Nonetheless, remote control is still incredibly useful in active setups, so its inclusion here, even as hardware, is appreciated.

And a minor point—the grilles are attached using magnets rather than pegs, screws, or grommets. This provides a neat appearance and allows an easy transition between listening to the speakers naked and protection from curious fingers and camouflage from living companions who don't appreciate the visual high-tech splendor of loudspeaker drivers.

Setup and Use

I placed the T1Es on rather inexpensive stands, with the vertical midpoint between tweeter and woofer at my ear level. After some experimentation with room position, the speakers ended up about 6' (1.8 m) apart, about 4' (1.2 m) from the side walls, and about 5' (1.5 m) from the back wall. My seating position was about 7' (2.1 m) back from the speakers. The tone controls were set to their center detent positions, and the EQ was likewise set to "flat." Toeing the speakers in slightly improved the imaging; I'd recommend having the axes cross just behind your preferred listening spot.

Besides my usual reference speakers, I also pressed some Kali LP-6s (see Resources) into comparative service. The Kalis, primarily designed for studio and mixing use, do not have anything close to the versatility of the T1Es, but are similarly priced and somewhat larger.

Initially, I had a problem—after using the speakers for a couple of weeks, they suddenly went silent. It turns out that a small percentage of the first production run (from which my samples were drawn) had a defective diode on the amplifier board. Reviewer's luck that I ended up getting one! In any case, for anyone who encounters this problem, Vanatoo offers to either replace the speaker or for those comfortable with electromechanical repair, sending the user a replacement amplifier. I opted for the latter, and doing the amp swap only took 15 minutes or so—the hardest part was getting the terminal clips off the woofer, since they are tight enough to give a gas-tight contact.

Once the amp was replaced, Bluetooth pairing was quick and completely free of adventure. Likewise, there were no issues using optical, coax,

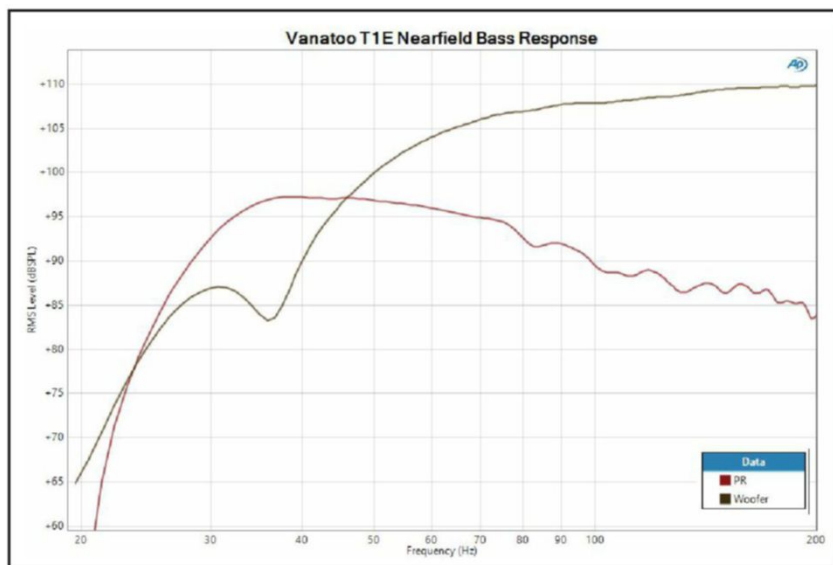


Figure 3: Near-field measurement of the woofer and passive radiator at low frequencies supports the subjective impression of strong and deep bass for a speaker of this size.

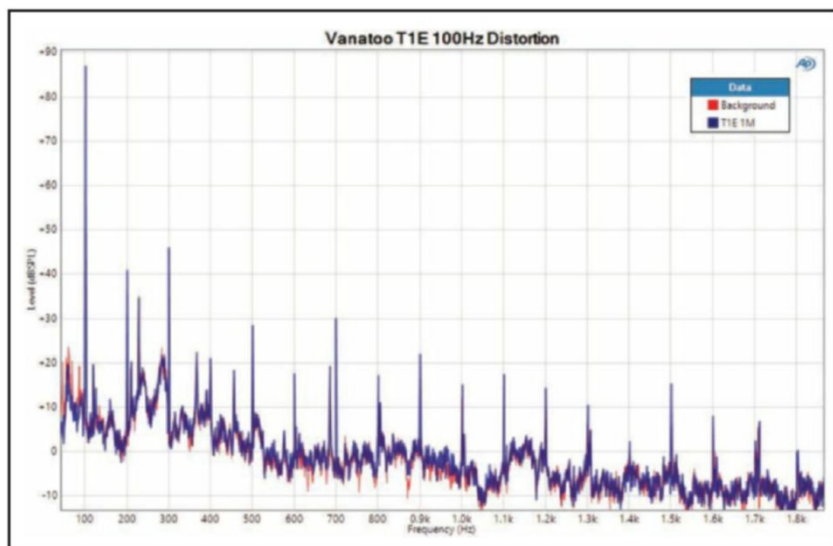


Figure 4: The spectrum of a 100 Hz sine wave shows notably low distortion for this size and price class.

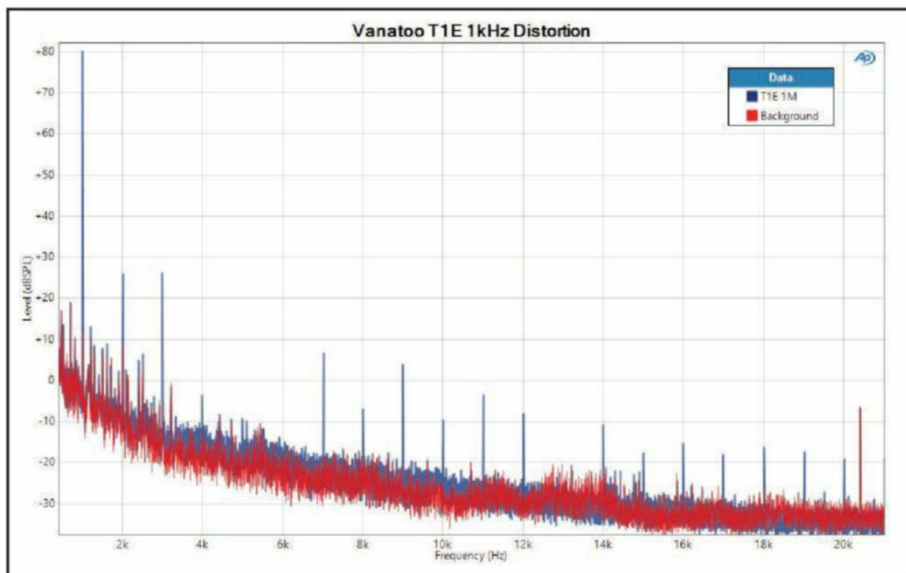


Figure 5: At 1 kHz, the T1E shows no tendency toward cone breakup or “cry” near the top of the woofer passband.

and analog inputs. At various times, I fed the T1Es from my laptop computer, my iPhone, and a couple of DACs I had sitting in my lab. All functioned flawlessly—people who want to avoid fuss and fiddling and just want to hear their tunes will be incredibly satisfied with the ease of getting things going and the input versatility. I was particularly struck by the simplicity of hooking up my iPhone and streaming from Amazon HD—basically, the entire array of components in a home stereo vanished, which may not impress audiophile friends, but is perfect for people who have music, not equipment, as front and center in their lives.

The reference material I use for evaluation is an assortment of different acoustic recordings (including my own) that span classical, traditional jazz, folk/Americana, bluegrass, and big band. I also dropped a few rock recordings in there to see if I could get the T1Es to choke up. And the answer is, my ears gave out first. In a small to moderate-size room (mine is about 14’ wide, 20’ long, with an 8’ ceiling), these speakers can play very loudly without obvious strain. I don’t know if the limiters were kicking in, but if they were, their effect was subtle.

My first impression of the T1Es was a very solid bass for their size. Not just solid, but well-defined in a way that the bass of a lot of mini monitors isn’t. Too many

mini monitors pursue giving an impression of bass without really delivering; the BBC LS3/5A was a progenitor of that genre, but that sort of midbass hump to fatten things up negatively affects the apparent texture and definition in the bottom three octaves. This is very much not the case here! The very bottom octave isn’t there, of course, but from 40 Hz up (and 40 Hz sounds deeper than you might think), the quality and quantity of bass belie the speakers’ size and price. I was particularly struck by how natural recordings of upright bass sounded—this is where most mini monitors show an extreme weakness, either bloating the sound or eviscerating it. Definitely a strong point of these speakers!

Imaging and soundstage were very, very good—not quite up to the spooky disappearing act of the Vanatoo T0s, but better than most speakers. Localization of instruments is tight and stable with no wandering. Compared to both my reference speakers and the Kali LP-6s, the listening window is a bit more restricted, with both tone and image degrading when I moved more than a couple of feet away from the center.

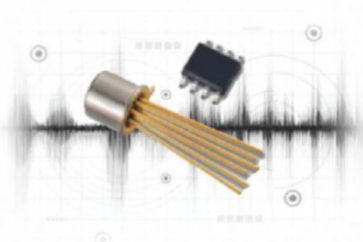
As far as tonality goes, human voices can be particularly revealing of midrange colorations, so I focused on those. In switching between the reference speakers and the Vanatoos, the T1Es sounded to me

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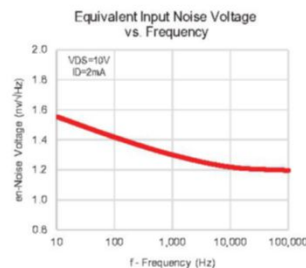


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slightly forward with a somewhat mellow but very clear upper treble.

To put things into perspective, these were rather small colorations, and perhaps the remarkable

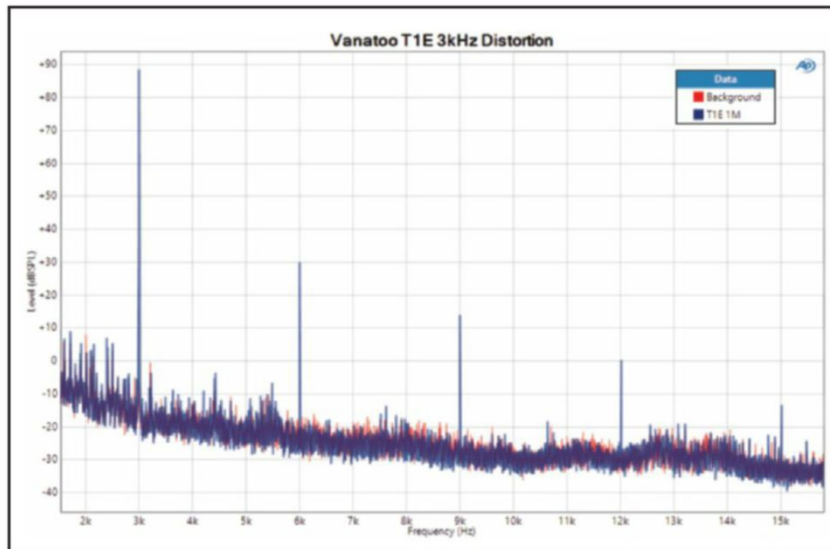


Figure 6: The 3 kHz distortion spectrum shows off the quality of the tweeter, even at large excursions.

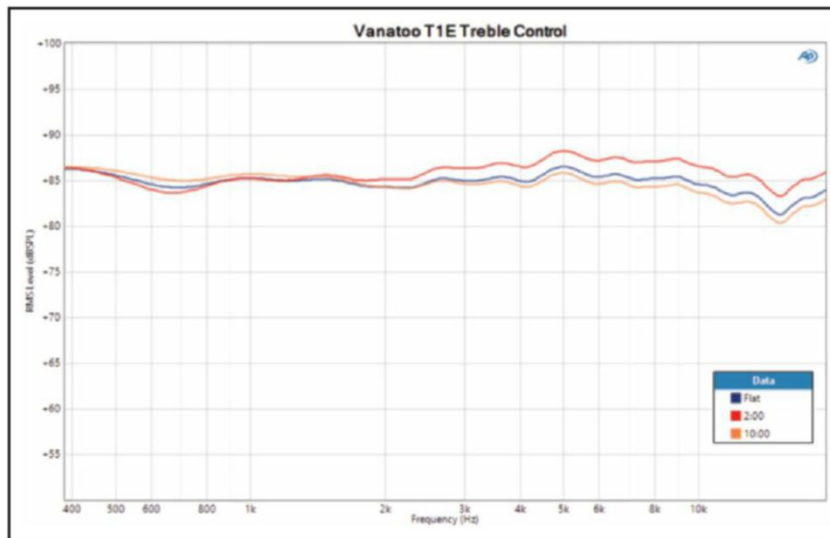


Figure 7: The treble control on the rear panel and the remote enables the user to easily make subtle adjustments of the tone.

Resources

D. Wiggins, "XBL: A Primer," Acoustic Development International, 2008, <http://www.acousticdev.com/Files/XBLPrimer.pdf>

S. Yaniger, "Vanatoo Transparent Zero Powered Speakers," *audioXpress*, January 2018, <https://www.audioxpress.com/files/attachment/2641>

S. Yaniger, "Kali Audio LP-6 Powered Studio Monitors," *audioXpress*, August 2019, <https://www.audioxpress.com/files/attachment/2689>

clarity of the sound enabled me to pick up on this. Indeed, the clarity and detail of acoustic music is what kept drawing me back, and I didn't feel a huge urge to get through the review and get back to my main system. By comparison, the Kali LP-6s seemed a bit more neutral, but also did not flatter any mediocre recordings, and of course, could only be driven by one of my balanced DACs.

For completeness, I also set up the T1Es on my desktop, fed by my lab computer. The EQ was reset to "Shelved," and the resulting sound was full-bodied and well-integrated. One concern for near-field listening like this is the noise from the power amplifiers; in the case of the T1Es, it was audible as a slight hiss, but it was significantly lower in volume than the hiss from the Kali LP-6s. And the Vanatoos took up significantly less desk space. I'd still prefer the Kalis for mixing and mastering, but for just plain music listening, I preferred using the T1Es.

Measurements

As usual, my measurement setup consisted of an Audio Precision APx525 audio analyzer, an Audio Precision APx1701 acoustic interface, and a PCB Piezotronics 376A32 1/2" calibrated condenser laboratory microphone. For near-field woofer and port measurements, I switched mics to the PCB Piezotronics 376A31 1/4" mic, which has a higher SPL capability than the 376A32. The APx525 optical output was connected via a TOSLINK cable to the T1E's optical input, with the T1E's volume control set to its midpoint. Frequency response and distortion measurements were acquired using a swept tone, which is transformed into an impulse response and Fourier transformed after gating out the reflections.

Figure 1 shows the quasi-anechoic on-axis frequency response, taken at six points and power averaged. The most noteworthy overall feature is the generally flat and neutral response up to about 10 kHz, but there's a notch near the 2 kHz crossover point and a bump at about 3.8 kHz. This may correlate with my impression of a forward sound. The top octave (10 kHz to 20 kHz) is shelved down by 4 dB to 5 dB, which is noticeable in direct comparison to my reference speakers and the Kali LP-6s, but also can make edgier recordings a bit more listenable. Nonetheless, I wish it weren't there and I could ameliorate poor recordings with the tone controls (see the Breaking News at the end of this review, though). On the positive side, there's no sign of a break-up resonance in the metal dome tweeter, nor anything similar for the woofer near the top of its passband.

The frequency response at 22.5° and 45° horizontally off axis (see **Figure 2**) may give some

clues about my observations on imaging and tonality. The response is very consistent at the smaller angle, which certainly was a positive as far as localization. At larger angles, things fall apart a bit, which may be related to the baffle size and geometry. I saw smoother behavior at those angles with the T0, perhaps because of the T0's smaller baffle and non-rectangular shape, and the T0 seemed to disappear more than most other speakers, including the T1E.

Figure 3 shows the near-field response of the woofer and the passive radiator. The 40 Hz tuning is evident, as is the strong output down to that frequency. The T1Es did not sound at all thin or bass-shy, as well as showing no bloatedness or fake midbass boosting, and here's data showing why.

The distortion spectra are also of interest. **Figures 4–6** show the distortion at 100 Hz, 1 kHz, and 3 kHz, respectively. The background room noise (dominated by the fans of the Audio Precision test gear) is shown in the red curves as a baseline. These aren't championship results, but they are still quite good, and for the price and size are extremely good. The 100 Hz distortion in particular is quite low for a driver this small. Note that the XBL technology seems to nearly eliminate fourth-order and higher even harmonics, and reduces the level of second harmonic, which in most conventional designs is the principal distortion component.

Figure 7 shows the effect of the treble tone control, with settings at the center detent (12:00), slightly turned down (10:00), and slightly turned up (2:00). At maximum and minimum, the response changes almost 10 dB, but as can be seen here, it's easy to get more subtle and tasteful adjustments.

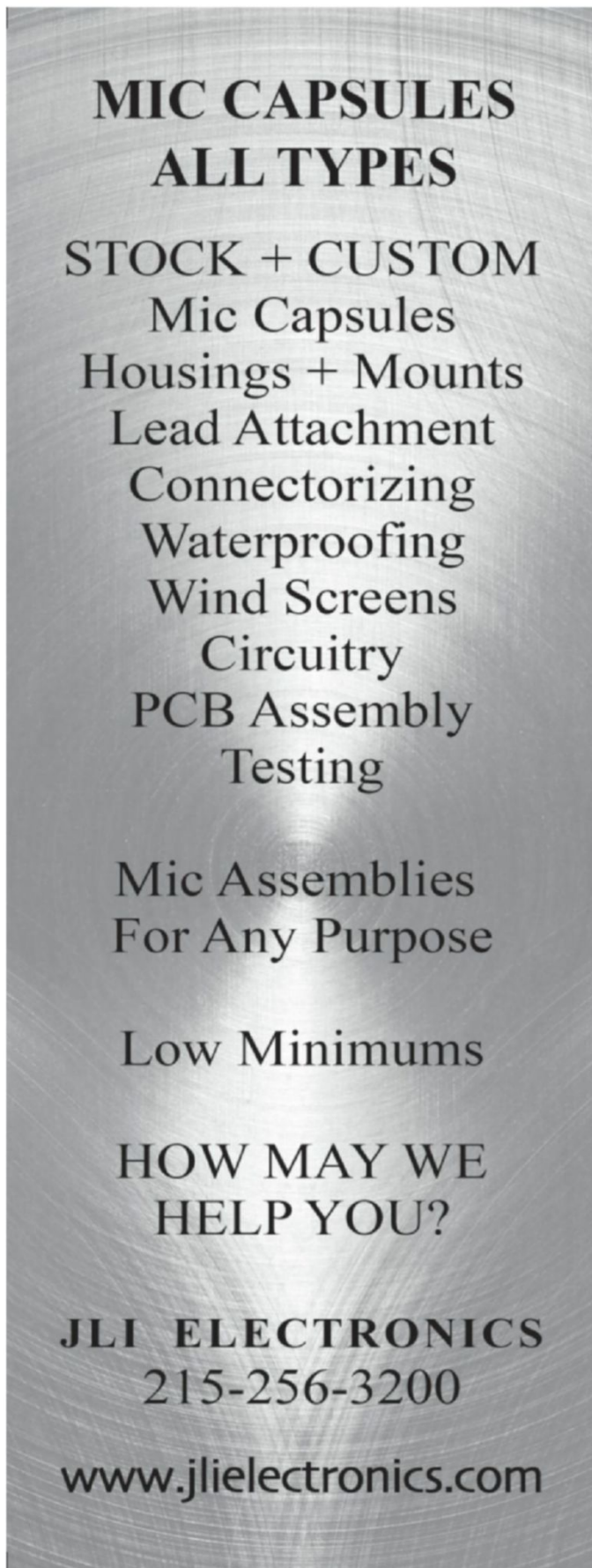
Overall, the measurements here are quite good and show both thoughtful engineering trade-offs and a few minor areas where improvements could be implemented in future updates.

Conclusions

After having several pairs of the latest generation of powered speakers cycling through my lab and living room, I am still amazed at what modern design and characterization tools, materials, and smart engineering can do to narrow the gap between state-of-the-art and state-of-the-affordable. Besides sounding great, the T1Es offer music lovers a simple and versatile package that can be easily configured to be used in rooms or on desktops, and fed by just about any imaginable source. If your priority is getting the music playing rather than fiddling with impressive equipment, the Vanatoo Transparent One Encore will more than satisfy you.

Breaking News

As of press-time, Vanatoo has released a firmware update (version 5.17), which is intended to fix a few of the minor frequency response issues I saw, as well as clean up some aspects of the polar pattern at larger angles. After some small adventures getting the firmware to load, the updates were implemented. My early impressions are that the new firmware fixed the two minor sonic issues I noted, the slight forwardness and the top octave droop. 🎧



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DIY Expert Enjoys the Challenges of Speaker Design

An Interview with Charlie Laub

By
Shannon Becker
(United States)



Charlie Laub designs loudspeakers because he loves the challenge of perfecting the sound.

SHANNON BECKER: How did you first get involved in loudspeaker design?

CHARLIE LAUB: I've been interested in loudspeakers since I was about 10 years old, when I rebuilt a pair of two-way "classic" bookshelf monitors with the help of my older brother and some trips to RadioShack. This was your standard fare of a 6.5" woofer and cone tweeter in a sealed box, with the single crossover component being a series cap on the tweeter. To my surprise, just rebuilding the speaker with new parts made a huge difference to the sound quality. The bar was low and I was not a critical listener at that age, but this was the hook that pulled me into the hobby and sparked my interest, and I remember it well to this day.

Over the next decade or so my interest in loudspeakers continued to grow but my involvement was limited to reading books such as *High Performance Loudspeakers* (by Martin Colloms) and an early version of the *Loudspeaker Design Cookbook* (by Vance Dickason). Later, in college, I sometimes went to the library to read back issues of the *Journal of the Audio Engineering Society*. I especially recall being captivated by an article about electronically assisted sixth-order vented alignments by Don Keele, Jr. and another article by Karl Erik Stahl about his "ACE-BASS" circuit that included feedback around the driver. Both of these articles describe how electronic filters or circuits could be

used to enhance the performance of the driver, and this sort of thing was soon a splinter in my mind that spurred me on to read about active filters and how they are used in loudspeaker crossovers.

BECKER: Tell us about your first DIY loudspeaker.

LAUB: Although by this point I had earned my BS in engineering (but not electrical engineering) I did not know a pole from a zero and I had never touched a soldering iron. Not knowing what I didn't know, I hatched a plan to build my first DIY loudspeaker—an active system using high-quality drivers and electronic crossovers. I bought a pair of D21 and D52AF from Dynaudio and a Meniscus brand 10" woofer. I had no tools and couldn't build a boxed cabinet so I decided I would try an "open baffle" speaker, consisting of a triangular-shaped baffle and open-backed woofer.

To build the crossover, I bought power supplies and a set of four electronic crossover boards from Marchand Electronics. I spent hours carefully assembling these into a 4U Budd Rackmount case that I had purchased. I also bought a multichannel amp by Rotel to drive the system. It seemed I had all the parts I needed, all top quality. What could possibly go wrong?

I had envisioned that I could put everything together and then adjust the crossover points to fine tune the sound. Veils would be lifted! Unfortunately

I had some serious holes in my understanding of how a loudspeaker radiates sound, how drivers sum, the effect of phase, and so on. Despite the invested time and expense, I found that the system just didn't sound all that good but I didn't really understand why. Undaunted, I thought that perhaps I just needed a subwoofer to augment the thin sounding bass. So I bought some 18" Eminence pro drivers, put each of them in a big bare MDF box, and bought a Linkwitz-Transform circuit from Marchand Electronics.

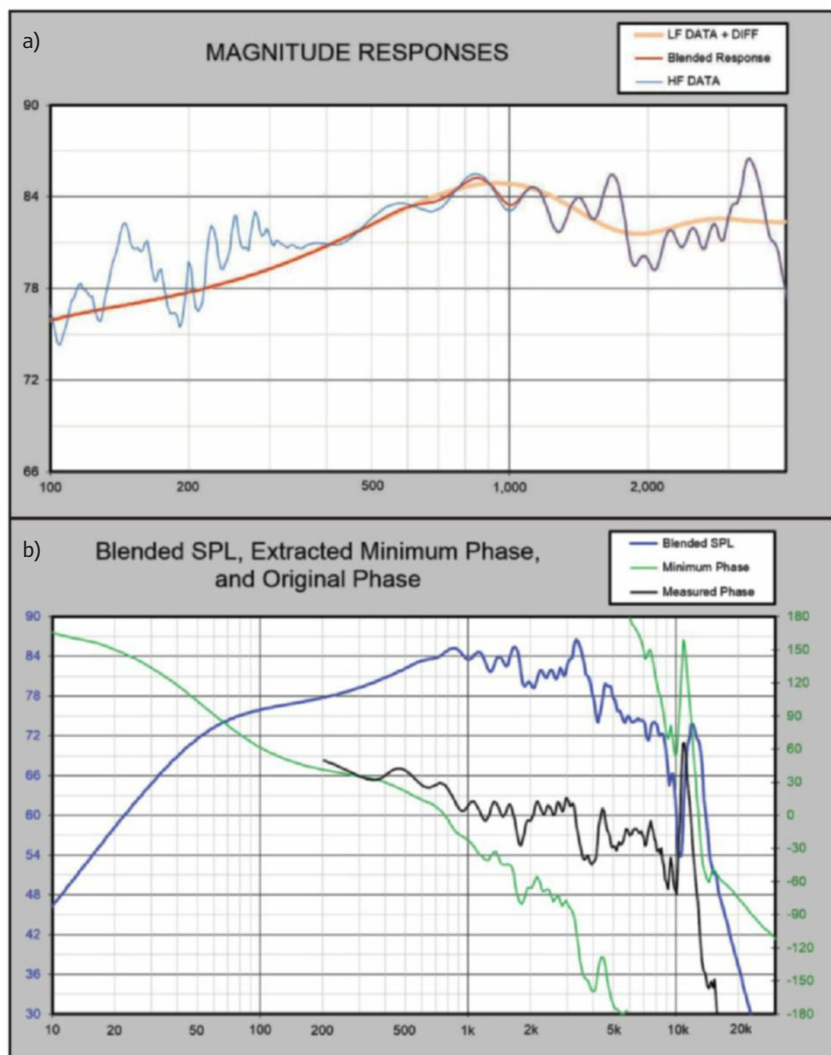
Again I found that this did not really result in the deep bass and amazing sounding loudspeaker that I had hoped for. It wasn't horrible, but it definitely didn't sound right, and flaws were audible on various recordings. After a couple of years' time I had to relocate and the system was dismantled. All that remains is one grainy picture of me in front of my mountain bike at the time, with one speaker in the background. Over the next several years I didn't build any speakers, but I kept on reading and learning about them.

BECKER: And yet, you tried again later with a DSP design?

LAUB: After another move or two I again had the urge to design and build a speaker. By this time (early 2000s) I was hearing about this thing called "DSP" and the hot item for DIYers back then was the Behringer DCX2496 electronic crossover. The sour taste from my earlier failed build was still in the back of my mind, so I decided that I should put my engineering knowledge to work and do some modeling of how the speaker and crossover, together, would produce sound.

I turned to Excel and started to create some spreadsheets that, over time, grew into a way to create a full and proper model of the loudspeaker and crossover as "seen" from a point in space (my Active Crossover Design Tools or "ACD"). I realized that I could take measurements with a microphone to obtain data about each driver at the "listening position" and then incorporate it into my model. This would be superior to using manufacturer datasheets, like I had done for my earlier speaker, because it would capture the driver, the box response, and the diffraction signature.

About this time, I discovered the measurement program ARTA, and bought my first measurement microphone and preamp. Now I could really get some accurate amplitude and phase data on the drivers, in box, and when combined with the crossover output would have a complete model of the loudspeaker.



Here are two screenshots of the FRD Blender software that Charlie Laub helped develop with fellow DIYer Jeff Bagby. Shown are the magnitude responses (a) and the blended vs. original responses (b).

My model of the crossover filters consisted of the transfer function of each filter in the Laplace operator(s). In Excel, I could perform all my calculations using complex numbers, and once I had stepped through the reproduction chain, the amplitude and phase were obtained. As a result of the study of filters and transfer functions, I realized there were filters that I could not implement on my DCX crossover.

At about this time I heard about a new DSP product from a company called "miniDSP" that offered an "advanced biquad" programming option. This meant that any filter response could be implemented, and this would give me great flexibility in designing the crossover. I immediately bought a pair of 2x4 from miniDSP, sold my DCX, and wrote a biquad transfer function coefficient calculator.

This is an open-baffle speaker system that Charlie built circa 2014.



BECKER: How did the FRD Blender come about?

LAUB: There was still one missing piece of the puzzle. I had a microphone and could take measurements, but always indoors. Why were all of these peaks and dips appearing in my measurement data? After some reading, I realized that this was reflections of the sound off of the walls that was mixing with the direct sound and causing interference. The solution was to use ARTA's gating function to eliminate these reflections from the time-record before obtaining the frequency response. But this often had the effect that much

of the low-frequency part of the spectrum was smoothed and the lowest frequency data was lost. I had read that you could take a near-field measurement and combine this with the higher frequency part of the far-field measurement, to get the wide-band response. Usually this was done by hard-splicing the two data sets together at some frequency. This seemed a little tricky to me, and sometimes the data did not combine smoothly using this approach. So I decided to work up another Excel spreadsheet that I could use to "blend" the band where the two datasets overlapped so that would smoothly combine.

This later became the "FRD Blender" and after talking it over with fellow DIYer Jeff Bagby, we decided to work together on the project. Jeff had already written some excellent tools for DIYers, including a box modeler and diffraction simulator. I had written the "blending" part of the tool, including a way to extract the minimum phase response from the resulting blended frequency using only Excel worksheet functions, so no external DLL library or routine was needed. Jeff realized that if my work could be combined with some of the aspects of his box modeler and diffraction simulator, it would make it possible for average DIYers to combine easy-to-make measurements into a wide band response that rivaled what you could get in an anechoic chamber. This resulted in the FRD Blender tool that many DIY loudspeaker builders use today.

BECKER: How did you get involved with the DIY audio community?

LAUB: With all of these tools, techniques, and measurement capabilities now within my grasp I could get back to the task to building loudspeakers. I built a couple of two-, three-, and four-way systems. Mostly I was not excited about the sound, and while they measured well, I just could not figure out why they seemed to sound flat and lack depth to the soundstage.

At this point I was living in northern California and started to run a DIY audio club. To provide some interest during a meeting I contacted Siegfried Linkwitz to come and give a talk to the members. This was one of several times he graciously gave his time to talk about loudspeakers, acoustics, and demo some of his own loudspeaker systems. I realized that Siegfried had an immense amount of information available on his website, and he was kind enough to answer an occasional question from me by email. I began to learn more about acoustics, psychoacoustics, and was able to listen to his Pluto, Orion, and later the LX521 loudspeaker.



This is the test fixture for 20 Hz-20 kHz dipole speaker development.

Over time, I found myself becoming more and more interested in open-baffle loudspeakers. This was not only from listening to Siegfried's systems but also after hearing a large DIY open-baffle speaker designed by John Busch and demoed at one of our DIY club meetings. I found the impact, sound stage, and general musicality of properly designed open-baffle systems to be much more satisfying than most boxed speakers, and I directed my own DIY efforts toward these type of systems. Since I had the tools and hardware to implement DSP crossover, and since open-baffle loudspeakers typically require more corrections and have more complicated crossovers, this seemed like a good way to combine all my interests into a single project.

BECKER: Tell us about your current projects.

LAUB: During the last couple of years, I have built several open-baffle loudspeaker prototypes using cone drivers. I have been paying close attention to the off-axis radiation, and how the diameter and design of a coned, moving coil driver influences the output, on and off axis, and especially to the rear.

My current goal is to try to create a 20 Hz to 20 kHz dipole loudspeaker—one that has identical front and rear radiation over the full audio band. This is much more than having an “open back” and even if they are relatively small or “full range,” most drivers are not truly dipolar above 1 kHz to 2 kHz because the magnet structure and the basket enclosure attenuate and/or filter the rear output. To get around this problem I will be using planar midrange and tweeter drivers in my next project about above 800 Hz.

I'm not adept at making beautiful cabinetry and have only a few tools and basic woodworking skills at my disposal, so often my projects are simply intended as a demonstration platform rather than a finished product, with no consideration to aesthetics. With each build I feel that I am getting closer to the end goal, and learning more and more all the time.

To make it possible to test out combinations of drivers “in the nude” (without a baffle) for my 20 Hz to 20 kHz dipole concept, I recently created a speaker “test fixture” consisting of widely spaced aluminum channel for vertical support and horizontal wooden cross members. A set of aluminum rods runs between holes in the upper cross members to provide convenient hanging points, and wires are used to hang the drivers so that they are decoupled from the structure and other drivers. By sliding the hanging points forward and back, I can physically align the acoustic centers of the drivers. The openness of the frame, and the lack of vibration trans-

mission help to keep the sound free of coloration and permits each driver stand on its own merits. Eventually, when I have the engineering done, I will work on a more polished and aesthetically pleasing way to put it all together.

BECKER: Do you have any advice for *audioXpress* readers who are thinking of building their own loudspeakers?

LAUB: There are lots of ways for even a novice to build a quality DIY loudspeaker. Speaker kits make this easy. These include the speaker drivers and all crossover parts. You build a cabinet based on the plans provided and install the drivers and crossover. Some kits even come with a pre-built cabinet. From there the rabbit hole can go as deep as you want it.

For the serious designer/builder you will need measurement equipment (like a calibrated microphone), audio measurement software, and crossover design software. There are excellent freeware versions of software available now, and a good measurement microphone costs less than \$100. Some woodworking equipment is needed, such as a router and circle cutting jig, and a table saw (even a small contractor saw) is very handy. Get in there and get your hands dirty building some small two-way speakers and then you can move the bar up to tackle more challenging projects. There are online DIY communities that can help you, and look for local DIY clubs and enthusiasts in your area.

BECKER: Finally, can you recommend any useful new parts or promising technologies to *audioXpress* readers? Anything that you're using now that you think readers should know about?

LAUB: During the past decade or so, the biggest advances for DIY loudspeakers enthusiasts have been increasingly affordable and capable measurement tools, software, and low-cost DSP electronic crossover units. Examples of these that are worth a look include USB microphones such as the UMIK-1 from miniDSP, the ARTA, and STEPS audio measurement and analysis programs from ARTA LABS, the loudspeaker design package VituixCAD by Kimmo Saunisto (<https://kimmosaunisto.net/Software/Software.html>), and DSP crossovers such as the t.racks 4x4 Mini from Thomann GmbH.

For me, DIY audio has really been about the journey—decades of trial and error, of learning and improving, of imagining and creating. For me, this has always been a challenging and fun pursuit and one that I know will continue for many years to come. 

Powering Your Circuits (Part 1)

Battery Power

Ed Simon has provided *audioXpress* with a six-part article series that explores the choices and considerations of using different power sources for audio circuits. In this first article, he discusses the positives and negatives of battery power use.

By
Ed Simon
(United States)

It takes power to make sound equipment work. This can come from batteries, standard Alternating Current (AC) power lines, other technologies such as solar power, or even other forms of harvested energy. The goal for an audio power supply is to provide a clean stable source of adequate current and voltage.

Batteries are used to power sound equipment when it is meant to be portable and sometimes to avoid any AC power line noise issues. Batteries are composed of two or more individual cells. Each cell typically produces a voltage of 1.2 V to 4.2 V typically. The more cells the higher the voltage. The physically larger each cell, the more energy stored for any selected chemistry, which often means higher current available.

Initial Considerations

The first consideration is: What voltage is required? Then how much current? What is the reasonable operating life? Are the batteries to be rechargeable or disposable?

The basic issues of voltage and current are based on the circuitry that is being powered. As batteries open circuit (without any load) have a definite upper voltage limit, it is often possible to run closer to the maximum voltage ratings of the electronic components than you could with an AC line-powered unregulated power supply. Of course, as batteries provide power their voltage will drop so they should only be used with circuits that can tolerate power rail drops.

A single battery or perhaps several may be considered. If you want to run an op-amp-based circuit with positive and negative voltage supply rails, the current drain is often the same for both rails so two of the same type of battery should work nicely.

If there is additional circuit load on, say, the positive rail, then the choice may be to use a separate battery for that load or to use a larger battery for the positive rail. If there is twice the current draw on the positive rail, you could use two batteries in parallel. This tends not to work very well, as any imbalance in the energy stored in the batteries will end up having the higher-state battery try to charge the lower one. As the balance will never be perfect, there will always be a small extra drain. Now this may not be an issue in large capacity batteries (e.g., the lead acid automotive type), but for small ones it may be.

Another method would be to use a larger capacity battery. As there are standard sizes, you can often find one that has the capacity needed. If properly matched, then all of the batteries powering the circuit will need to be replaced at the same time.

The final decision will often be determined by what kinds of batteries are available and the costs.

Battery Life

The current drawn by the circuit will affect battery life. A typical 9 V alkaline battery has rating of 400 mAh to 600 mAh (milliampere hours.) Now, if you try to draw 200 mA from the

battery, it will not last two hours! But if you draw 20 mA it will probably last the 20 hours and if you draw 5 mA it may last far longer than 120 hours! That is because the useful capacity of any battery is dependent on the actual load current. Operating at too high or low temperature also reduces most batteries' capacity.

Sometimes, placing a capacitor across the battery will increase the service life. That is because the capacitor (or supercap) will provide short term peak currents and let the battery see a more consistent, lower current load. It will also lower the AC source impedance of the power supply as the load frequency rises. This may help prevent interstage feedback through the power rails. (Sometimes this is a low-frequency oscillation called motor-boating due to the sound it makes.)

Occasionally there is a desire to use a single battery to provide a bipolar power supply. The technique to do this is to establish a "virtual ground." This is usually set at one half of the batteries voltage. The simplest method is to use a resistive divider (see **Figure 1**). This is often used when it is desired to use a low power op-amp from a single voltage supply.

As this method provides ground impedance equal to one half of the resistor value plus a bit from the internal battery impedance, it is often too high. An op-amp may be added to reduce the current drain and provide a stiffer reference. At the same time higher value resistors may be used to lower the current drain. The output current capacity of the op-amp will be the limit to this technique.

If frequently used one may prefer rechargeable batteries. Most rechargeable batteries do not have the same energy storage capacity as regular types of the same size. Even worse with use or age their total energy storage capacity decreases. So when designing for a rechargeable battery, good practice is to design for a lower than rated load capacity, allowing a longer total battery life (see **Figure 2**).

One common error is to use the same number of cells for the rechargeable battery as for a disposable one. This would work if the open-circuit voltage is the same, but that is not a common occurrence.

The Internal Impedance

A major advantage of batteries is low power supply noise. It turns out that the noise of batteries can never be lower than the equivalent thermal noise of their internal impedances, but it can approach that! The noise of batteries is so low that heroic efforts are required to measure it. So for practical purposes we just look at the internal impedance of them.

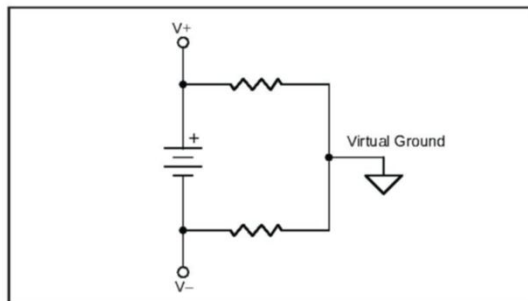


Figure 1: Occasionally, there is a desire to use a single battery to provide a bipolar power supply. The simplest method is to use a resistive divider.

To get the internal impedance of a battery, all you do is measure it lightly loaded (10% of design value is a good test) and then again with a resistor that will draw the same current as your design needs. Take the voltage drop from light load to test load and divide that by the increase in current drawn. That is the effective internal resistance.

To determine the expected noise you can use the simplified formula for noise voltage:

$$V_n = \sqrt{(1.62E-20 \times R \times Bw)} \text{ at room temperature (70°F)}$$

where Bw is the bandwidth of interest, often 20,000 Hz.

A handy number to remember is that a 62 Ω resistor is very close to 1 nV for a bandwidth of 1 Hz. The other handy number is 3,086 Ω ! That value resistor over the audio bandwidth of 20,000 Hz will produce 1 μ V of noise. If you want to do a quick calculation of any resistor value for noise over the audio band all you have to do is take the square root of that value after it is divided by 3,086. That will give you the noise in microvolts.

As an example, a 100 k Ω resistor would be $\sqrt{(100,000/3086)}$ or 5.69 μ V. Keep in mind that as this is average noise, the instantaneous value could be much higher. This is the long-term equivalent value.

If you parallel resistors, the noise level will drop. Placed in series, it will increase. Keep in mind the square root function is involved, so placing two of the same value resistor in series the noise does not double, it goes up by the square root of 2 or is 1.414 times the noise of a single unit. Placed in parallel, identical resistors will have the noise decrease to 0.707 of the original value.

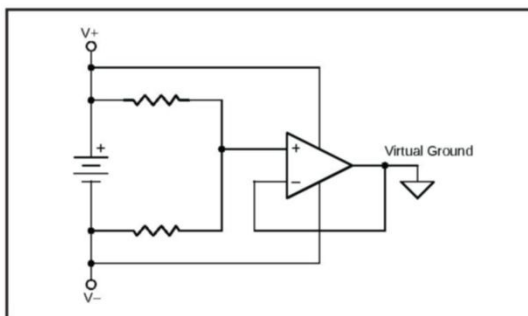


Figure 2: A better virtual ground uses an op-amp to lower the impedance.

Battery Type	Ohms (Ω) Provided
12 V, 7 AH lead acid battery	0.19 Ω
4 $\times$ 3.2 V, 2.3 AH FeLiPo (Lithium) w/protection circuit	0.11 Ω
10 $\times$ NiMh AA cells, 12.8 V, 700 mAh	1.75 Ω
8 $\times$ Alkaline D cells, 12.66 V	1.70 Ω
4 $\times$ 3.2 V, 0.4 AH FeLiPo AA size	0.94 Ω
8 $\times$ Alkaline AA cells	1.84 Ω

Table 1: These are the internal resistances for some typical batteries.

As a rule, higher current capacity cells in batteries will have lower internal resistance, resulting in lower noise. Typical low impedance batteries are nickel metal hydride, alkaline, lead acid, and the lithium ion cells made for power tools (lithium iron phosphate) not the ones intended for memory back-up use.

Table 1 shows typical battery internal resistances. These were done with a simple set-up. A 100 Ω resistor was paralleled with a 10 Ω resistor that could be switched in or out.

Noise Reduction

Batteries contribute little noise to the circuits they power, but they do something else quite important. They reduce noise on the power rails! If you have a circuit that draws 10 mA at 12 V, it may be modeled as a 1,200 Ω resistor. (The circuit being powered may actually produce more noise than a plain resistor but never less.) The resistor would have an audio band noise level of about 620 nV. If placed in parallel with the lithium battery tested, it would require a capacitor of at least 100,000 μ F to reduce the low-frequency noise to this level.

It should come as no surprise that batteries also have capacitance, as anything with two plates has some capacitance. In addition there is an issue of float voltage. When some batteries, particularly lead acid ones, are kept charged, it is often at a higher voltage than in normal use. So an initial current drain will result in a larger voltage drop as the current begins to be drawn.

Connectors

Another consideration with batteries is the type of connector used. As most batteries require a mechanical connection, the connectors should exert good contact force. The connections need to be clean of corrosion. A poor connection will show up not just as an increase in power supply noise, but can also cause a voltage drop that will limit power and reduce battery life. Many rechargeable batteries are available with welded tabs or friction fit connectors.

Charging

A consideration with the lithium batteries is that charging and discharging must be carefully controlled. A special circuit is often used to control these battery packs. It is designed to prevent the cells from ever being overcharged as that damages them. Discharging them too much also does them in. The result is that the protection circuit very slightly increase, the effective internal resistance and noise of lithium battery packs.

Battery Disposal

A major issue is that lithium batteries are considered hazardous and require special handling and shipping. So as a rule of thumb, for a given physical size nickel cadmium, nickel metal hydride, or some types of lithium batteries have low noise. Larger gel type lead acid batteries are a good choice where space permits. Alkaline batteries pretty much are standard for disposable types.

It should not be a surprise that batteries degrade with age. So even with oversized cells you may find you need to change rechargeable batteries every year or three to keep the low noise performance and load life.

Contamination

A less common issue with batteries is contamination. Small impurities during manufacturing may cause bubbles or spots where the chemistry is not uniform. These may create excess noise. In older liquid lead acid cells, dirt would often get in and fall to the bottom of the acid reservoir. With more modern gelled chemistry, any impurity tends to stay in place. So batteries by themselves may not be the ultimate power supply.

An interesting issue is actual battery plate size. These plates may act as an antenna to pick up line and other radiated noise! So batteries may actually work better for you when placed in a shielded enclosure.

Next month, we will investigate the second most popular method of powering sound equipment, which is to convert the AC power line into useful lower voltage direct current (DC).

About the Author

Ed Simon has a BSEE from Carnegie Mellon University. He was employed by Cameradio, a regional electronics parts distributor from 1973-1975. In 1975, Ed started Edward Simon & Co. He has published more than 40 articles and taught courses for the NSCA, Carnegie Mellon University and the University of Washington among others. His company specializes in sound and communication systems, systems for public facilities, houses of worship, and allied engineering. The business has completed more than 400 church installations, and 150 other installations including shopping malls, teleconferencing facilities, performing arts centers, television studios, and radio stations.



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Effect of Load Impedance on the Output of Hollow-State Power Amplifiers

In this article, our hollow-state electronics expert tells us why the effect of load resistance on maximum amplifier output power is different for hollow-state vs. solid-state amplifiers.

By
Richard Honeycutt
(United States)

The effect of load resistance on maximum amplifier output power is very different for hollow-state vs. solid-state amplifiers. A solid-state amplifier acts pretty much as a pure voltage source: Its output voltage is virtually unaffected by the load impedance. A hollow-state amplifier's load impedance is reflected to the plate circuit of the output tubes by the transformer, having a direct effect on their gain, and a much smaller effect on the output-stage distortion. The magnitude of these effects depends upon the amount of negative feedback and its type: series or shunt feedback.

Another characteristic that differs between hollow- and solid-state amplifiers is the effect of the output terminals being left open or shorted together. A solid-state amplifier will try to produce a given output voltage into the load, resulting in more and more output current as the load impedance is decreased. Shorted loads would always lead to crispy output transistors were it not for the universal use of current-limiting circuits in solid-state power amps. Open loads, on the other hand, are no problem for solid-state amplifiers. Because of the effect of load impedance on output-stage gain, shorted loads do not cause overcurrents in hollow-state amplifiers. Open loads, however, can cause severe clipping, which can create large voltage spikes that can result in arc-over, damaging output transformers and output tubes.

Speaker Impedance

Note that, despite the common parlance, we are not discussing "impedance matching" here, except in the sense that the speaker impedance may or may not match the rated impedance of the amp's output terminal. That rated impedance is not the same as the amplifier's (Thevenin) output impedance. Neither hollow-state nor solid-state amps ever drive loads whose impedance matches the internal impedance of the amp. Despite years of faulty teaching by uninformed science teachers, the maximum power transfer resulting from having matched source and load impedances in almost any audio circuit comes with serious disadvantages. For solid-state power amplifiers, these disadvantages include smoke and perhaps flame. For hollow-state amplifiers, the output impedance depends to a certain extent upon the connected load impedance, and so matching the amp's output impedance to a load impedance would be a very contrived effort unlikely to optimize any aspect of amplifier performance.

The Damping Factor

When we think of power amplifiers as controlled sources, we often consider them to be Thevenin (voltage) sources. A Thevenin source contains an ideal voltage source in series with a Thevenin (internal) impedance. This Thevenin impedance is the output impedance of the amp. For most solid-

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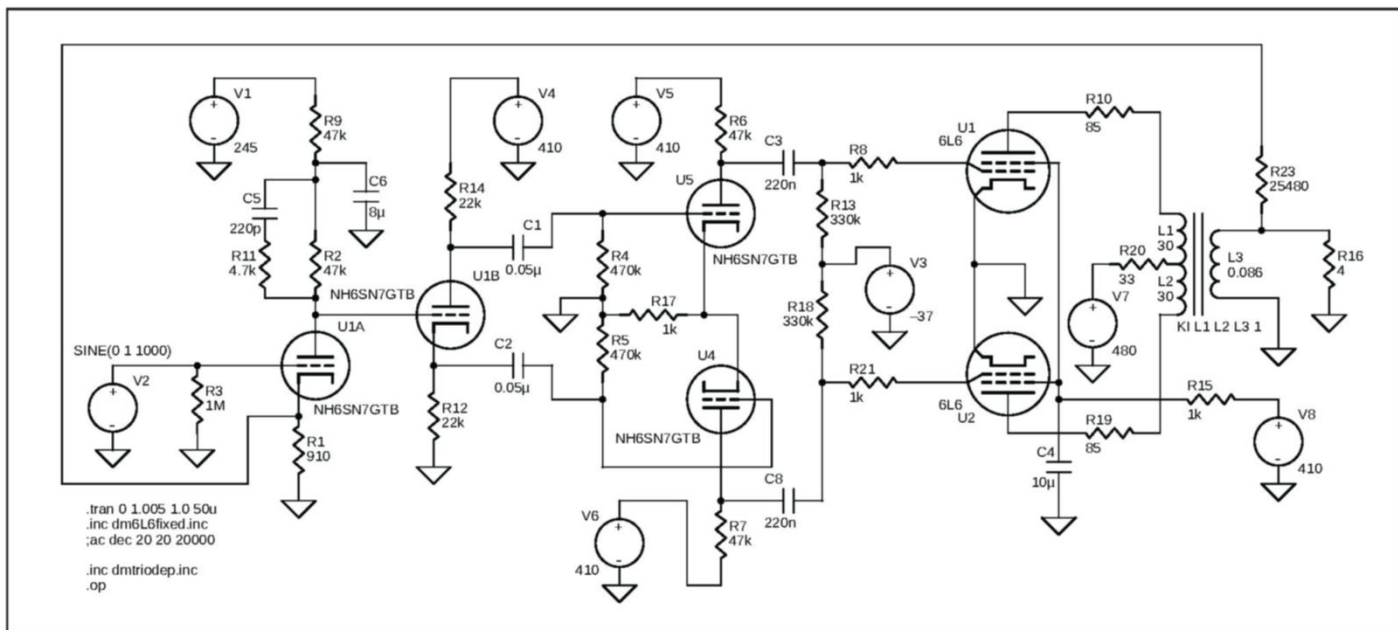


Figure 1: This LTSpice model of a typical hollow-state power amplifier was used for the simulations.

state amplifiers, it is near zero ohms. The output impedance of a hollow-state amp depends upon the negative feedback. The ratio of the nominal load resistance of an amplifier to the resistance of the load is called the damping factor. This name came about because of the following facts:

- A speaker voice coil moving in a magnetic field produces a voltage.
- That voltage, when connected across the output resistance of the amplifier, causes power to be dissipated in the output resistance.
- Removing power from the acoustical/mechanical/electrical circuit is the effect of resistance.
- A resistance connected in series with a reactive circuit (e.g., a speaker) produces mechanical damping—in this case, damping the motion of the voice coil.

High damping factors improve the transient response of a speaker/amplifier system by preventing the speaker cone from continuing to “ring” after the drive voltage from the amplifier ceases. It might, therefore, be assumed that

a higher damping factor is always better, but as Howard Tremaine pointed out in his book *Audio Cyclopaedia*:

Damping factor = $Z_{LS}/(Z_{out}+R_{vc})$, where Z_{LS} is the nominal impedance of the speaker, Z_{out} is the output impedance of the amplifier, and R_{vc} is the voice-coil resistance of the speaker.

Since R_{vc} is always in the circuit, an 8 Ω speaker cannot achieve an effective damping factor above about 20 in most cases. The very low output impedance of solid-state amplifiers causes them to have very high damping factors, but the damping factors of hollow-state amps may be much lower. Some hollow-state amplifiers had potentiometers that controlled damping factors within the stated range of about 0.1 to 100. However, as Tremaine mentioned, the actual range was nearer 0.1 to 20. In the days when “bass-reflex” (vented) speaker designs were less well-understood than they are today, “boom-box speakers” were common, and amplifier damping could make a substantial difference in the transient response of an audio system.



Figure 2: The power amp produced this output wave into a 16-Ω load.

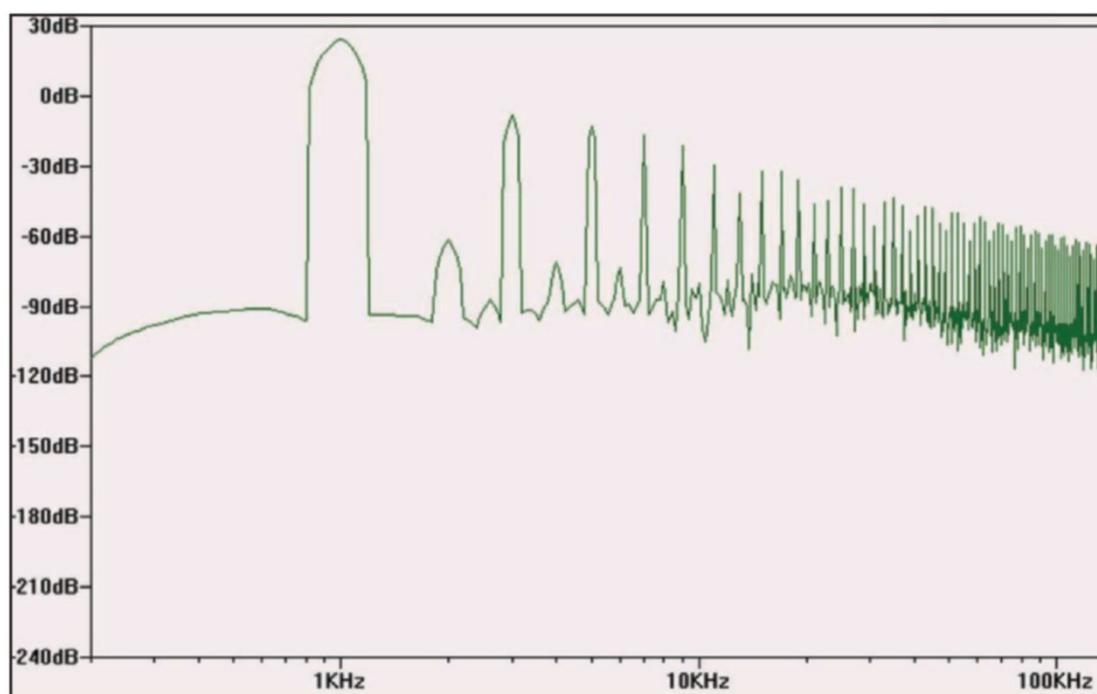
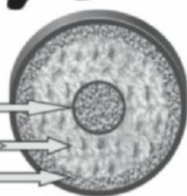


Figure 3: The FFT of the output wave into a 16- Ω load shows the fundamental at 1 kHz, plus harmonics at multiples of 1 kHz.

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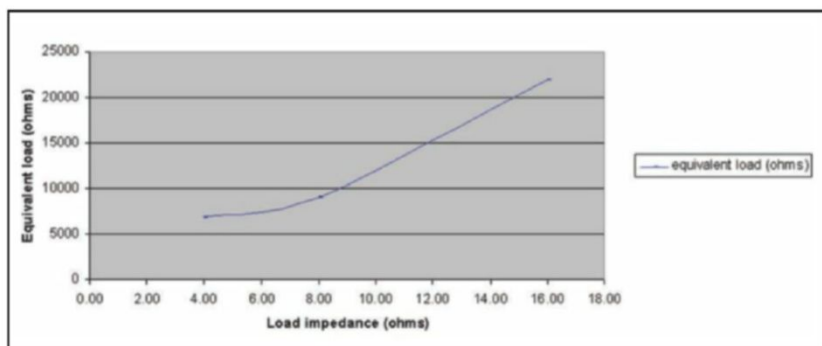


Figure 4: The output transformer reflects a load-dependent plate-to-plate impedance.

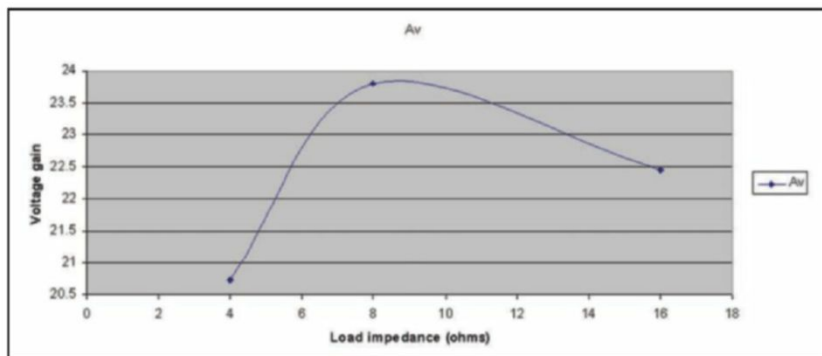


Figure 5: The amp's voltage gain varies according to the load impedance.

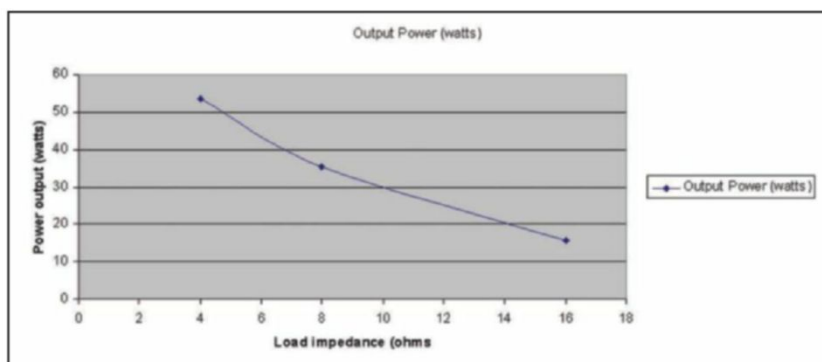


Figure 6: The output power also varies according to the load impedance.

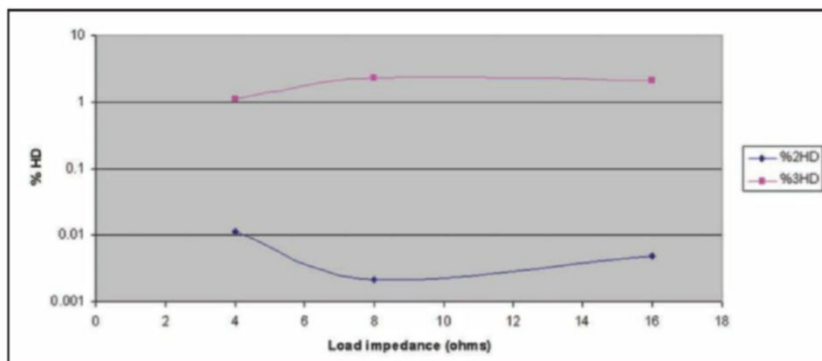


Figure 7: Second and third harmonic distortion vary only slightly with the load impedances.

Practical Examples

As an example, **Figure 1** shows an LTspice model of a typical hollow-state power amplifier using a 6L6GC push-pull pair and 6SN7 voltage amplifier and phase splitter. I ran simulations showing the performance of this amplifier using a 1 V_p sine-wave input, with three separate loads: 4 Ω, 8 Ω, and 16 Ω. Note that the output transformer has only one output—there are no separate taps for other impedances.

The output wave produced into a 16 Ω load is shown in **Figure 2**. Note that the wave is clipped at 22.45 V_p. **Figure 3** shows a Fast Fourier Transform (FFT) of the output wave. Because of the negative feedback, the output wave shows very little distortion before the onset of clipping, as you may have noticed in Figure 2. Simulated output waves for 4 Ω and 8 Ω loads showed very similar waves.

In order to compare the effects of different loads on the amp performance, I calculated performance parameters for each load impedance. As mentioned earlier, the load impedance affects the plate-to-plate load seen by the push-pull pair, affecting voltage gain and output power. **Figure 4** shows the magnitude of the plate-to-plate load impedance plotted against the resistance of the load.

The RCA specifications for a 6L6GC show a recommended plate-to-plate load of 5600 Ω for a Class AB1 push-pull pair with a 450 V B+, close to the modeled 480 V B+. Note that the actual value for this circuit is a bit higher than 5,600 Ω for a 4 Ω load, and considerably higher for 8 Ω and 16 Ω loads.

How does this change affect the gain of the amplifier? **Figure 5** shows the voltage gain for each of the three load impedances. The voltage gain is defined as output voltage divided by input voltage, and in this case, the output voltage was taken across the output transformer secondary. As expected, the amp's overall voltage gain depends greatly upon the load impedance.

And how does the change in voltage gain translate into output power? **Figure 6** shows the output power for the three load impedances. As we would expect, maximum output power is produced when the plate-to-plate load is nearest the value recommended on the spec sheet.

Not surprisingly, the distortion is very low, because of the negative feedback. As shown in **Figure 7**, changes in load impedance have only a small effect on harmonic distortion.

Connecting a speaker instead of a dummy load resistor will introduce some frequency-dependent effects, due to the fact that a speaker presents a complex impedance: voice-coil resistance plus

capacitive and inductive effects. You will notice in **Figure 8** that the equivalent circuit of a speaker includes inductance and capacitance as well as resistance, and is not a simple circuit. The impedance magnitude of this circuit varies significantly with frequency, which will affect the amplifier's voltage gain and output power in a frequency-dependent way.

Conclusions

So what does all this mean? Of course, if we expect a hollow-state amplifier to perform according to the manufacturer's specifications, we should use a speaker having the rated impedance. But using a speaker of a different impedance is not likely to damage the amplifier, although it will cause it to produce a different output power than is specified.

Many hollow-state amps designed for home listening have a variety of transformer taps to allow for speakers of various impedances. Musical instrument amps, on the other hand, usually have only one output, commonly designed for a 4 Ω or 8 Ω nominal impedance. These amps are often used with speaker stacks of perhaps two to four cabinets, each having an impedance of 4 Ω to 16 Ω . Mutual

coupling of the speakers increases the efficiency of these systems, so any decreased output power due to a lower impedance is more than compensated. **ax**

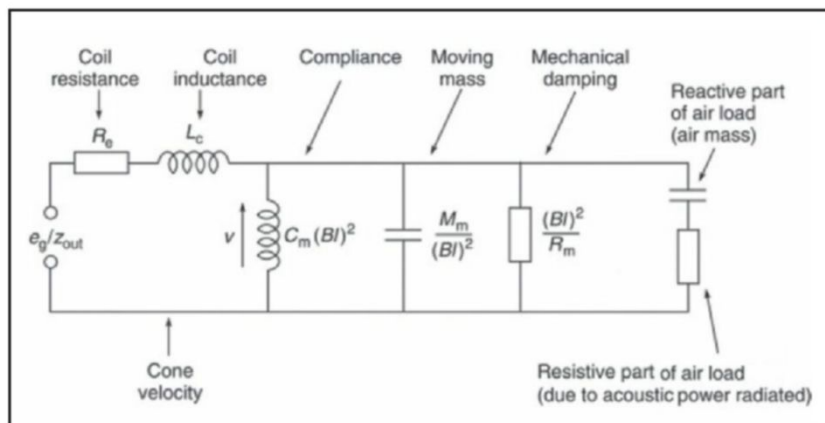


Figure 8: The equivalent circuit of a speaker is a complex LRC circuit. (Image courtesy of audio judgement)

Resources

H. Tremaine, *Audio Cyclopedia*, 2nd Edition, Howard W. Sams, Indianapolis, 1969.

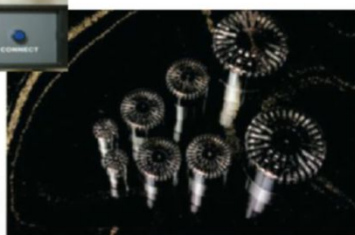
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Salon Audio Montréal/Montréal Audiofest

Hotel Bonaventure, Montréal, Canada
montrealaudiofest.org



The Montréal Audiofest is now steadily on a growth track since the troubling events of 2016, when the former Montréal Audio Show promoted by the Chester Group was canceled just two weeks before the scheduled days. Fortunately, Michel Plante and Sarah Tremblay, the local original promoters of this event with more than 30 years of tradition, quickly revived the project calling for the industry's support to make the show dates at the original hotel and on the scheduled dates. Just one minor change, the event was renamed from "Show" to "Fest" and is now the Salon Audio Montréal/Montréal Audiofest. Three years after that small miracle, the 2019 edition was considered the best ever and the Montréal Audiofest is now firmly part of the high-end audio calendar. In 2020, Montréal Audiofest is celebrating 33 years and will take place once again at the Bonaventure Hotel, which has just received a refreshing facelift.

March 31–April 5, 2020

Prolight+Sound Frankfurt

April 1–4

Musikmesse

Messe Frankfurt Exhibition

Frankfurt, Germany

www.musikmesse.com

www.prolight-sound.com

www.messefrankfurt.com



In 2019, the world of entertainment technology and music business gathered once again at Prolight+Sound and Musikmesse in Frankfurt. With a revised hall layout, Messe Frankfurt responded to all the important demands from exhibitors and visitors by locating the entire Audio, Sound + Recording product segment in Hall 8.0—the biggest ground-level exhibition hall at the constantly expanding Frankfurt Fair and Exhibition Centre. As a result, the pro audio section of Prolight+Sound was once again a vibrant trade show, even if with lots of empty spaces and "fill-in" themed areas that fooled no one.

At the end of the show, the feeling was mixed. Some exhibitors were extremely happy with the visitors' quality and the business they conducted. Others, said they would not return. In the completely new Hall 12, where the world of show lighting and stage technology was located, the feeling was completely different. It was "party like it's 1999" and most of the exhibitors probably never even noticed that there was something happening in the other 11 halls of the gigantic Messe Frankfurt complex. Most probably, that area of Prolight+Sound will continue to expand. Just a pity that the two "lighting" and "sound" areas aren't closely located to each other...

For 2020, Prolight+Sound celebrates 25 years, and *audioXpress* and *Voice Coil* will be represented there once again, because it is the show where we can meet with almost all the important driver companies from Italy, Germany, France, UK, Spain, and even some from the US. Visitors can also meet with some of the largest pro audio companies, such as L-Acoustics, d&b audiotechnik, and RCF under the best possible conditions, and even listen to the latest PA systems actually playing in the Live Sound Arena outdoors (when it's not raining).

At the other extreme of the Messe complex, there is still a show called Musikmesse, which is now a sort of musical festival starting on Wednesday and extending to Saturday. But in place of the vibrant trade show where retailers from all over the world would arrive excited to try out the latest gear and place orders, there are now lots of curtains, empty carpets, and the weird silence of an art gallery.

According to Frankfurt Messe, the 2020 edition of Prolight+Sound will offer new trade-visitor services, an even bigger gamut of conferences, more demonstration rooms for elaborate product presentations, and there is a Hosted Buyer program: Messe Frankfurt will bear flight and hotel costs for a limited number of new top buyers, which are selected in cooperation with key-account exhibitors.



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