

Acoustics Today



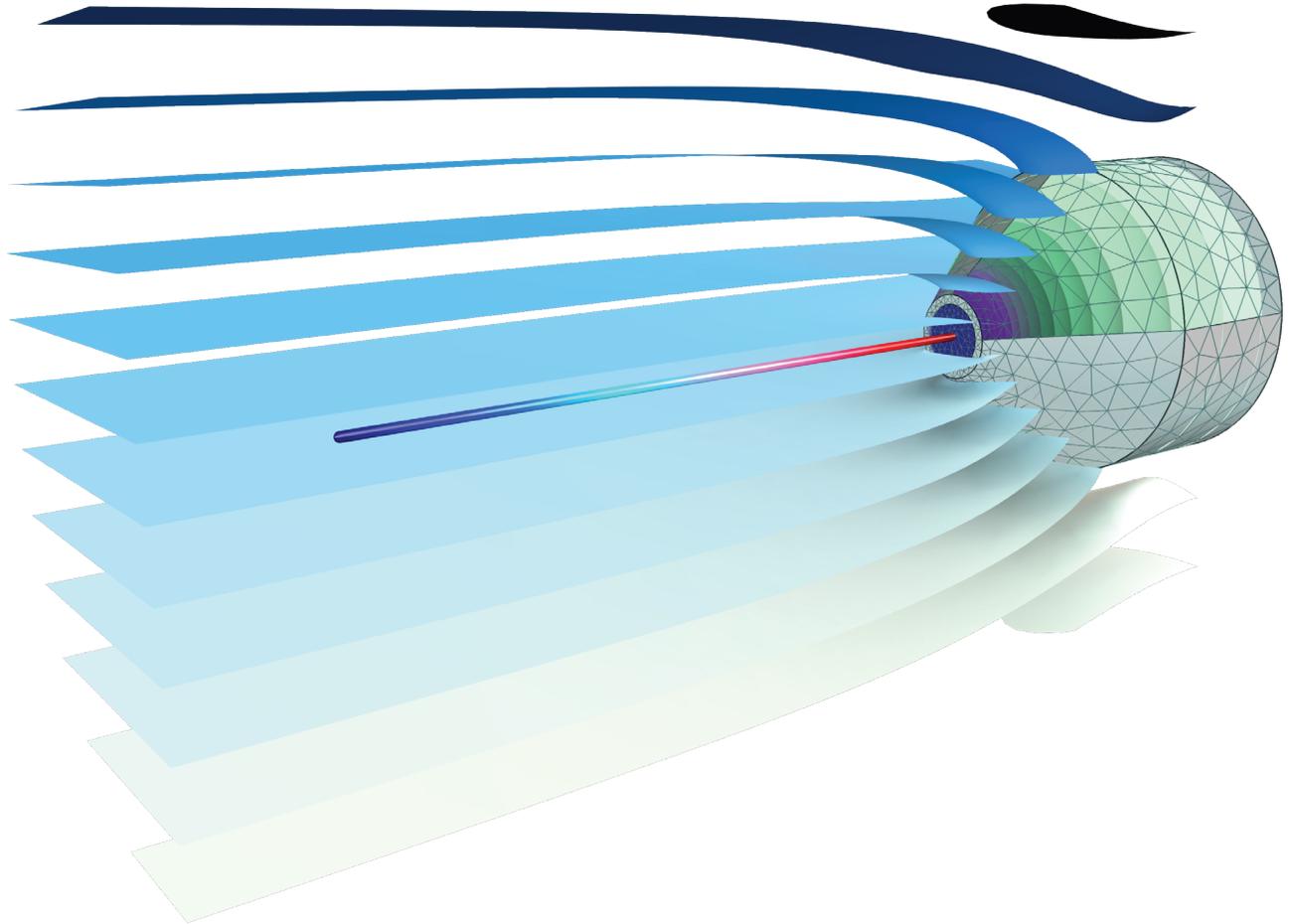
A Publication of the Acoustical Society of America

The Remarkable Cochlear Implant

Also In This Issue

- Defense Applications of Acoustic Signal Processing
- Snap, Crackle and Pop: Theracoustic Cavitation
- The Art of Concert Hall Acoustics: Current Trends and Questions in Research and Design
- Too Young for the Cocktail Party?
- Heptuna's Contributions to Biosonar

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

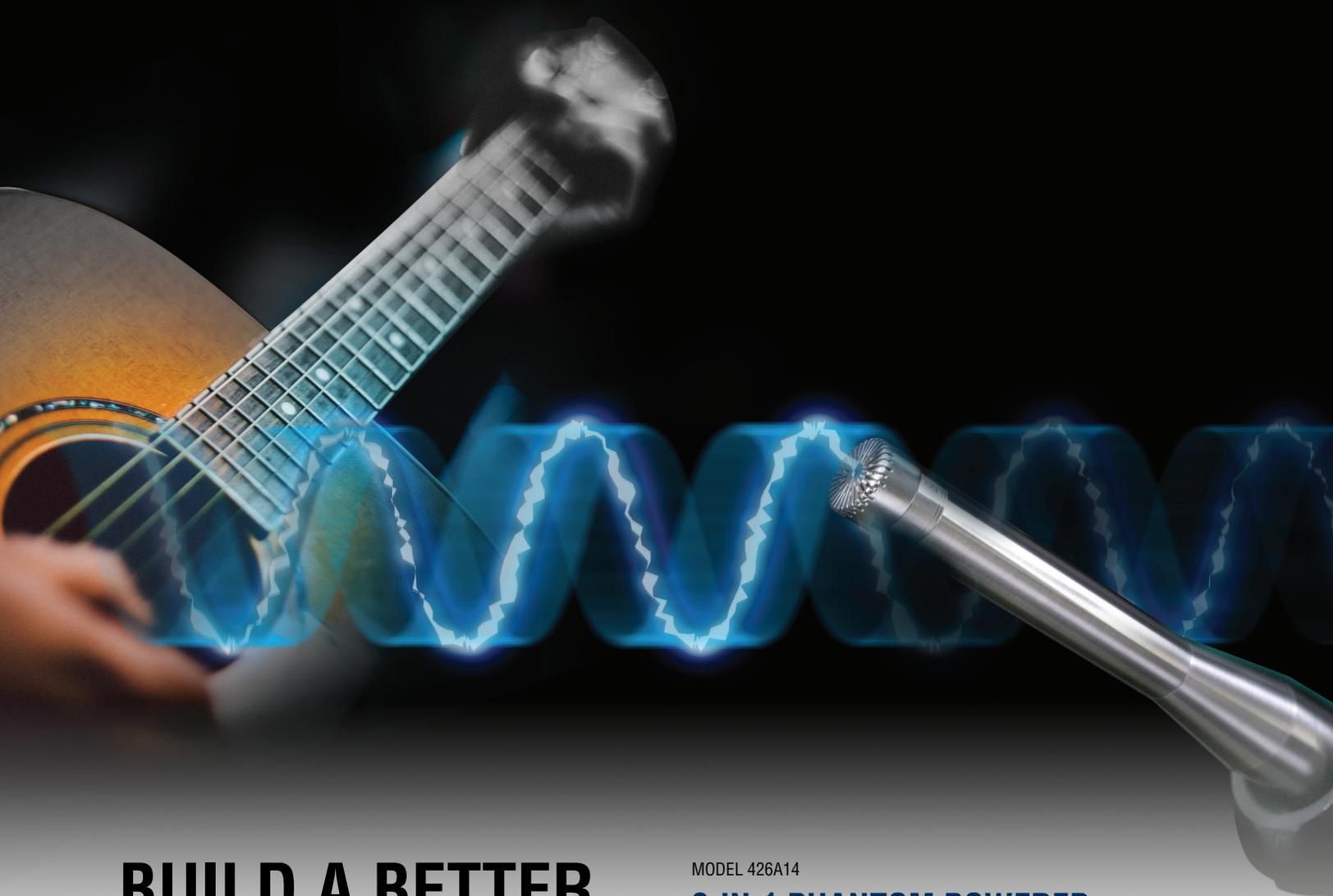


Visualization of the total acoustic pressure field around and inside an elastic probe tube extension attached to a microphone case.

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

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

comsol.blog/cocktail-party-problem



BUILD A BETTER INSTRUMENT WITH A BETTER INSTRUMENT.

MODEL 426A14

2-IN-1 PHANTOM POWERED MICROPHONE PREAMPLIFIER

- 156 dB 1% THD and flat frequency response for high definition recording and modeling instruments
- Quickly change between ½" and ¼" O.V., IEC 61094-4 compliant microphones
- Fast rise times for superior impulse responses and transients



 **PCB PIEZOTRONICS**
MTS SYSTEMS CORPORATION

1 800 828 8840 | pcb.com/phantom

MTS Sensors, a division of MTS Systems Corporation (NASDAQ: MTSC), vastly expanded its range of products and solutions after MTS acquired PCB Piezotronics, Inc. in July, 2016. PCB Piezotronics, Inc. is a wholly owned subsidiary of MTS Systems Corp.; IMI Sensors and Larson Davis are divisions of PCB Piezotronics, Inc.; Accometrics, Inc. and The Modal Shop, Inc. are subsidiaries of PCB Piezotronics, Inc.

**MTS**
SENSORS

Acoustics Today

 **ASA** A Publication of the Acoustical Society of America

TABLE OF CONTENTS

- 6** From the Editor
7 From the President

Featured Articles

- 10** Defense Applications of Acoustic Signal Processing –
Brian G. Ferguson

Acoustic signal processing for enhanced situational awareness during military operations on land and under the sea.

- 19** Snap, Crackle and Pop: Theracoustic Cavitation
 – *Michael D. Gray, Eleanor P. Stride, and Constantin-C. Coussios*

Emerging techniques for making, mapping, and using acoustically driven bubbles within the body enable a broad range of innovative therapeutic applications.

- 28** The Art of Concert Hall Acoustics: Current Trends and Questions in Research and Design – *Kelsey A. Hochgraf*

Concert hall design exists at the intersection of art, science and engineering, where acousticians continue to demystify aural excellence

- 37** Too Young for the Cocktail Party? – *Lori J. Leibold, Emily Buss, and Lauren Calandruccio*

One reason why children and cocktail parties do not mix.

- 44** Heptuna's Contributions to Biosonar – *Patrick Moore and Arthur N. Popper*

The dolphin Heptuna participated in over 30 studies that helped define what is known about biosonar.

- 53** The Remarkable Cochlear Implant and Possibilities for the Next Large Step Forward – *Blake S. Wilson*

The modern cochlear implant is an astonishing success; however, room remains for improvement and greater access to this already-marvelous technology.

Sound Perspectives

- 62** Awards and Prizes Announcement

- 63** Ask an Acoustician – *Kent L. Gee and Micheal L. Dent*

- 66** Scientists with Hearing Loss Changing Perspectives in STEMM – *Henry J. Adler, J. Tilak Ratnanather, Peter S. Steyger, and Brad N. Buran.*

- 71** International Student Challenge Problem in Acoustic Signal Processing 2019 – *Brian G. Ferguson, R. Lee Culver and Kay L. Gemba*

Departments

- 76** Obituary – *Jozef J. Zwislocki | 1922 – 2018*

- 80** Classifieds, Business Directory, Advertisers Index

About The Cover



From “The Remarkable Cochlear Implant and Possibilities for the Next Large Step Forward” by Blake S. Wilson. X-ray image of the implanted cochlea showing the electrode array in the scala tympani. Each channel of processing includes a band-pass filter. Image from Hüttenbrink et al. (2002), Movements of cochlear implant electrodes inside the cochlea during insertion: An x-ray microscopy study, *Otology & Neurotology* 23(2), 187-191, <https://journals.lww.com/otology-neurotology>, with permission.

With GRAS acoustic sensors, you can trust your data regardless of the application.

Only GRAS offers a complete line of high-performance standard and custom acoustic sensors ideal for use in any research, test & measurement, and production applications. Our microphones are designed for high quality, durability and performance that our R&D, QA, and production customers have come to expect and trust.

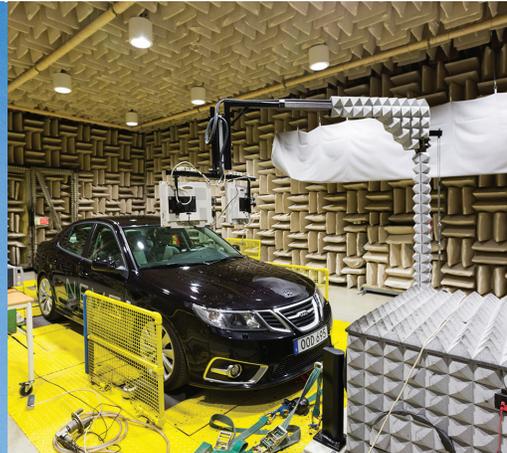
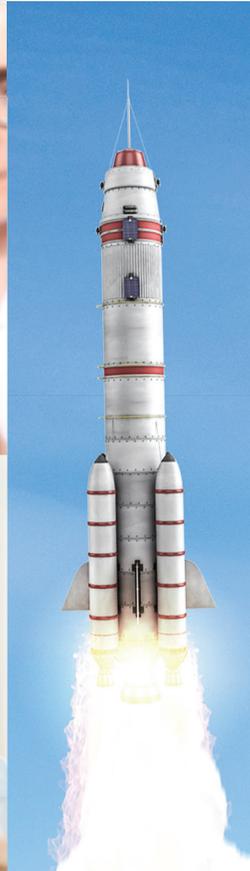
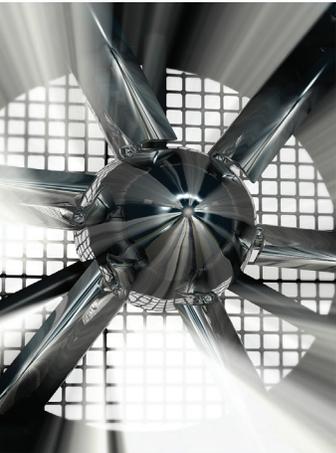
Contact GRAS today for a free evaluation of the perfect GRAS microphone for your application.



GRAS

Acoustic Sensors

- > Measurement microphone sets
- > Microphone cartridges
- > Preamplifiers
- > Low-noise sensors
- > Infrasound sensors
- > High resolution ear simulators
- > Microphones for NVH
- > Head & torso simulators
- > Test fixtures
- > Custom designed microphones
- > Hemisphere & sound power kits
- > Calibration systems and services



P: 800.579.GRAS
E: sales@gras.us

www.gras.us

Editor

Arthur N. Popper | apopper@umd.edu

Associate Editor

Micheal L. Dent | mdent@buffalo.edu

Book Review Editor

Philip L. Marston | marston@wsu.edu

ASA Publications Staff

Mary Guillemette | maryguillemette@acousticstoday.org

Helen Wall Murray | helenwallmurray@acousticstoday.org

Kat Setzer | ksetzer@acousticalsociety.org

Helen A. Popper, *AT Copyeditor* | hapopper@gmail.com

Acoustics Today Intern

Gabrielle E. O'Brien | eobrien3@uw.edu

ASA Editor In Chief

James F. Lynch

Allan D. Pierce, *Emeritus*

 Follow us on Twitter @acousticsorg

Please see important *Acoustics Today* disclaimer at www.acousticstoday.org/disclaimer.

Acoustical Society of America

Lily M. Wang, *President*

Scott D. Sommerfeldt, *Vice President*

Victor W. Sparrow, *President-Elect*

Peggy B. Nelson, *Vice President-Elect*

David Feit, *Treasurer*

Christopher J. Struck, *Standards Director*

Susan E. Fox, *Executive Director*

ASA Web Development Office

Daniel Farrell | dfarrell@acousticstoday.org

Visit the online edition of *Acoustics Today* at AcousticsToday.org



Publications Office

P.O. Box 809, Mashpee, MA 02649

(508) 534-8645

Acoustical Society of America

The Acoustical Society of America was founded in 1929 “to increase and diffuse the knowledge of acoustics and to promote its practical applications.” Information about the Society can be found on the Internet site: www.acousticalsociety.org.

The Society has approximately 7,000 members, distributed worldwide, with over 30% living outside the United States.

Membership includes a variety of benefits, a list of which can be found at the website:

www.acousticalsociety.org/membership/asa-membership

All members receive online access to the entire contents of the *Journal of Acoustical Society of America* from 1929 to the present. New members are welcome, and several grades of membership, including low rates for students and for persons living in developing countries, are possible. Instructions for applying can be found at the Internet site above.

Acoustics Today (ISSN 1557-0215, coden ATCODK) Spring 2019, volume 15, issue 1, is published quarterly by the Acoustical Society of America, **Suite 300, 1305 Walt Whitman Rd., Melville, NY 11747-4300**. Periodicals Postage rates are paid at Huntington Station, NY, and additional mailing offices. POSTMASTER: Send address changes to *Acoustics Today*, Acoustical Society of America, **Suite 300, 1305 Walt Whitman Rd., Melville, NY 11747-4300**. Copyright 2019, Acoustical Society of America. All rights reserved. Single copies of individual articles may be made for private use or research. Authorization is given to copy articles beyond the use permitted by Sections 107 and 108 of the U.S. Copyright Law. To reproduce content from this publication, please obtain permission from Copyright Clearance Center, Inc., 222 Rosewood Drive, Danvers, MA 01923, USA via their website www.copyright.com, or contact them at (978)-750-8400. Persons desiring to photocopy materials for classroom use should contact the CCC Academic Permissions Service. Authorization does not extend to systematic or multiple reproduction, to copying for promotional purposes, to electronic storage or distribution, or to republication in any form. In all such cases, specific written permission from the Acoustical Society of America must be obtained. Permission is granted to quote from *Acoustics Today* with the customary acknowledgment of the source. To reprint a figure, table, or other excerpt requires the consent of one of the authors and notification to ASA. Address requests to AIPP Office of Rights and Permissions, **Suite 300, 1305 Walt Whitman Rd., Melville, NY 11747-4300**; Fax (516) 576-2450; Telephone (516) 576-2268; E-mail: rights@aip.org. An electronic version of *Acoustics Today* is also available online. Viewing and downloading articles from the online site is free to all. The articles may not be altered from their original printing and pages that include advertising may not be modified. Articles may not be reprinted or translated into another language and reprinted without prior approval from the Acoustical Society of America as indicated above.

Sound and Vibration Instrumentation

Scantek, Inc.



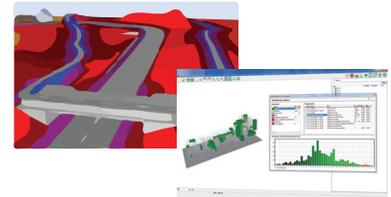
Sound Level Meters

Selection of sound level meters for simple noise level measurements or advanced acoustical analysis



Vibration Meters

Vibration meters for measuring overall vibration levels, simple to advanced FFT analysis and human exposure to vibration



Prediction Software

Software for prediction of environmental noise, building insulation and room acoustics using the latest standards



Building Acoustics

Systems for airborne sound transmission, impact insulation, STIPA, reverberation and other room acoustics measurements



Sound Localization

Near-field or far-field sound localization and identification using Norsonic's state of the art acoustic camera



Monitoring

Temporary or permanent remote monitoring of noise or vibration levels with notifications of exceeded limits



Specialized Test Systems

Impedance tubes, capacity and volume measurement systems, air-flow resistance measurement devices and calibration systems



Multi-Channel Systems

Multi-channel analyzers for sound power, vibration, building acoustics and FFT analysis in the laboratory or in the field



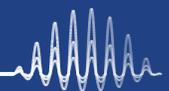
Industrial Hygiene

Noise alert systems and dosimeters for facility noise monitoring or hearing conservation programs

Scantek, Inc.

www.ScantekInc.com

800-224-3813





We are starting a new feature in our “Sound Perspectives” section that will appear in each spring and fall issue of *Acoustics Today* (AT), a list of the various awards and prizes that

will be given out and the new Fellows who will be honored by the Acoustical Society of America (ASA) at the spring and fall meetings (awardees and Fellows in this issue of AT will be honored at the Spring 2019 Meeting). The idea for including this list came from ASA President Lily Wang, and I thank her for her suggestion because inclusion of this list further enhances the ways in which AT can contribute to our Society.

I am particularly pleased that several of the awardees have been, or will be, AT authors. In fact, one author in this issue was just elected a Fellow. And I intend to scour these lists for possible future authors for the magazine.

I usually enjoy the mix of articles in the issues of AT, and this one is no exception. The first article by Michael D. Gray, Eleanor P. Stride, and Constantin-C. Coussios provides insight into the use of ultrasound in medical diagnosis. I invited Constantin to write this article after I heard him give an outstanding talk at an ASA meeting, and the article reflects the quality of that talk.

This is followed by an article written by Brian Ferguson on how signal processing is used in defense applications. Brian gives a great overview that includes defense issues both in the sea and on land and he presents ideas that, admittedly, I had never known even existed.

In the third article, Kelsey Hochgraf talks about the art of design of concert halls. Many readers might remember that Kelsey was the first person featured in our “Ask an Acoustician” series (bit.ly/2D4RkmI). After learning about Kelsey and her interests from that column, I invited her to do this article that gives a fascinating insight into a number of world-renowned concert halls.

The fourth article is by Lori Leibold, Emily Buss, and Lauren Calandrucchio. They have one of the more intriguing titles, and their article focuses on how children understand sound in noisy environments. Anyone with kids (or, in my case, grandkids) will find this piece interesting and a great complement to the article on classroom acoustics in the fall 2018 issue of AT bit.ly/2D4ydJt.

I will admit some “prejudice” to the fifth article. I was in San Diego about 18 months ago and had lunch with my friend, and former student, Patrick Moore. We started to reminisce (we go back many decades) and talked about a mutual “friend,” a dolphin by the name of Heptuna. As you will discover, Heptuna was a unique Navy “researcher.” After discussing the very many projects in which Heptuna participated, I invited Patrick to write about the history of this animal. He turned this on me and invited me to coauthor, and I could not resist. I trust you will see why!

The final article is by Blake Wilson, the only member of the ASA to ever win a Lasker Award (bit.ly/2AGBQnc). Blake is a pioneer in the development of cochlear implants, and he has written a wonderful history of this unique and important prosthetic that has helped numerous people with hearing issues.

As usual, we have a range of “Sound Perspectives” articles. “Ask an Acoustician” features Kent Gee, editor of *Proceedings of Meetings on Acoustics* (POMA). I have the pleasure of working with Kent as part of the ASA editorial team, and so it has been great to learn more about him.

This issue of AT also has a new International Student Challenge Problem that is described by Brian Ferguson, Lee Culver, and Kay Gemba. Although the problem is really designed for students in a limited number of technical committees (TCs), I trust that other members of the ASA will find the challenge interesting and potentially fun. And if other TCs want to develop similar types of challenges, I would be very pleased to feature them in AT.

Our third “Sound Perspectives” is, in my view, of particular importance and quite provocative. It is written by four researchers who are hearing impaired: Henry J. Adler, J. Tilak Ratnanather, Peter S. Steyger, and Brad N. Buran. I’ve known the first three authors for many years, and when I heard about their interests in sharing issues about being a hearing-impaired scientist and how they deal with the world, I immediately invited them to write for AT.

I am particularly pleased that one of the authors of this article is Dr. Buran. Brad was an undergraduate in my lab at the University of Maryland, College Park, working on the ultrastructure of fish ears. You can learn more about his career path in auditory neuroscience in the article. Having Brad in

Continued on page 9



Recent Actions on Increasing Awareness of Acoustics

The Fall 2018 Acoustical Society of America (ASA) Meeting in Victoria, BC, Canada, held in conjunction with the 2018

Acoustics Week in Canada, was a resounding success! Many thanks to Stan Dosso (general chair), Roberto Racca (technical program chair), the local organizing committee, the ASA staff, and the leadership of the Canadian Acoustical Association for their efforts in making it so.

As usual, the Executive Council of the ASA met several times to conduct the business of the Society. Among the items that were approved by the Executive Council were the selection of several recipients for assorted Society medals, awards, prizes, and special fellowships (see page 62 in this issue of *Acoustics Today* for the list); the election of new Fellows of the Society (acoustic.link/ASA-Fellows); a new five-year publishing partnership between the ASA and the American Institute of Physics Publishing; improvements to the budgeting process of the Society; and changes to the rules of the Society with regard to (1) the new elected treasurer position and (2) establishing a new membership engagement committee. The latter three items, in particular, are outcomes of strategies from the 2015 ASA Strategic Leadership for the Future Plan (acoustic.link/SLFP).

In the past two From the President columns in *Acoustics Today*, I've discussed recent actions related to two of the four primary goals from that strategic plan: membership engagement and diversity and financial stewardship. In this column, I'd like to summarize the progress we've made on one of the other main goals dealing with the awareness of acoustics: "ASA engages and informs consumers, members of industry, educational institutions, and government agencies to recognize important scientific acoustics contributions." A task force of ASA members has worked diligently with the ASA leadership and staff toward this goal, increasing the impact of the outreach activities of the Society. The efforts have successfully led to

- 1) expansion of the promotion of ASA activities and resources through emerging media and online content;
- (2) advancing the web and social media presence of the ASA;

- (3) improving the image of the ASA through a strategic branding initiative;
- (4) fostering members' ability to communicate about science;
- (5) considering how the Society should engage in advocating for public policy related to science and technology through government relations; and
- (6) increasing awareness and dissemination of ASA standards.

I won't be able to expound on all these initiatives in great detail here, but I will highlight some of the achievements to date.

One of the first items completed shortly after the Strategic Leadership for the Future Plan was released was the hiring of ASA Education and Outreach Coordinator Keeta Jones. Keeta has done an outstanding job championing and overseeing many ASA outreach activities. Visit exploresound.org to find a current compilation of available resources, including the updated Acoustics Programs Directory. If you teach in an acoustics program that isn't already represented in this directory, please be sure to submit an entry at acoustic.link/SubmitAcsProgram. Keeta also reports regularly about ASA education and outreach activities and programs in *Acoustics Today* (for examples, see acoustic.link/AT-ListenUp and acoustic.link/AT-INAD2018).

Another team member who has played a large role in improving the impact of the outreach activities of the ASA is Web Office Manager Dan Farrell. Our updated ASA webpage is a wonderful modern face for the Society along with the new ASA logo that was rolled out in 2018. Both Dan and Keeta have worked together to increase the presence of the ASA on social media as well, summarized in a recent *Acoustics Today* article (acoustic.link/AT-ASA-SocialMedia). I admit that I personally was not an early adopter of social media, and even now I am reticent about posting personal items on social media. However, I participated in a workshop in 2011 that taught me how social media can be leveraged to communicate our science more effectively to the broader public. I now believe strongly that having an active presence online and on social media (e.g., Twitter, Facebook, LinkedIn, YouTube) is a good tactic for the Society to spread the word about the remarkable work done by the Society and its many members. Please consider joining one or more of the ASA social media groups to help us increase dissemination further.

From the Editor

Continued from page 6

the lab was a wonderful experience, partly because he has a rather “wicked” sense of humor, but mostly because he helped everyone in my lab better understand and appreciate the unique role that hearing plays in our lives and the implications of hearing loss. Working with Brad was a tremendous learning experience for all of us (and I think for Brad as well). I encourage every member of the ASA to read this article and think about what the authors are saying. In many ways, all of us in acoustics can learn about the importance of sound from these four exceptional scholars.

Finally, I want to encourage everyone to think about using articles from *AT* for teaching purposes. To date, there are over 250 articles (and many essays) in past issues of *AT*, all of which are available online and open access. There is sufficient material in many areas where an instructor could assign a number of *AT* articles to their classes, either as part of course packs or by just giving the URLs. Indeed, if anyone does put together sets of articles for various classes, feel free to share the list (with the URLs) with me and it will be published in *AT* and/or placed on our website for the use of other instructors.

PAC International

World Leader in Noise Control Solutions



PACPRO™
RSIC

www.pac-intl.com



CLASSIFIED
UL US

RC-1 Boost



UL
CERTIFIED
SAFETY US-CA
R1.6638

RSIC-1

STC and IIC up to 70



CLASSIFIED
UL US

RSIC-SI-CRC EZ



CLASSIFIED
UL US

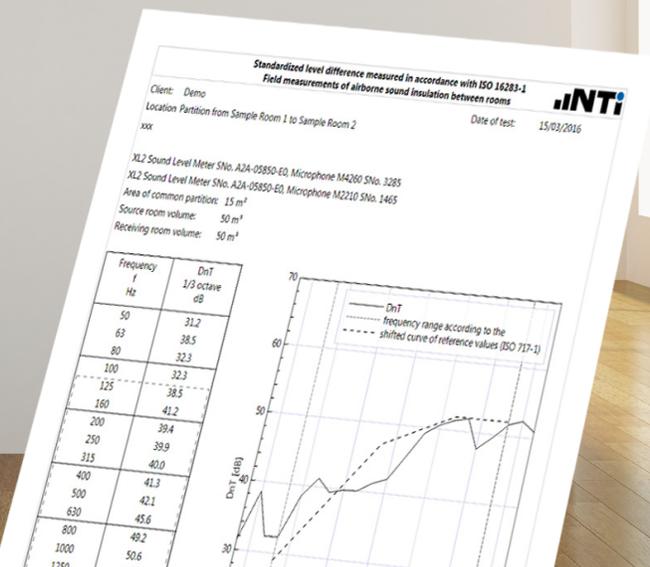
RSIC-SI-1 Ultra

RSIC® is a registered Trade Mark of PAC International, LLC. © PAC International, LLC. All Rights Reserved. • (866) 774-2100 •

Building Acoustics Test Solution







Frequency f Hz	DnT 1/3 octave dB
50	31.2
63	38.5
80	32.3
100	32.3
125	38.5
160	41.2
200	39.4
250	39.9
315	40.0
400	41.3
500	42.1
630	45.6
800	49.2
1000	50.6
1250	50.6

www.nti-audio.com/XL2

NTi Audio Inc., Tigard, Oregon, US
P: 0503 684 7050 E: americas@nti-audio.com

Complies with ASTM Standards

Defense Applications of Acoustic Signal Processing

Brian G. Ferguson

Address:

Defence Science and Technology (DST)
Group – Sydney
Department of Defence
Locked Bag 7005
Liverpool, New South Wales 1871
Australia

Email:

Brian.Ferguson@defence.gov.au

Acoustic signal processing for enhanced situational awareness during military operations on land and under the sea.

Introduction and Context

Warfighters use a variety of sensing technologies for reconnaissance, intelligence, and surveillance of the battle space. The sensor outputs are processed to extract tactical information on sources of military interest. The processing reveals the presence of sources (*detection process*) in the area of operations, their identities (*classification or recognition*), locations (*localization*), and their movement histories through the battle space (*tracking*). This information is used to compile the common operating picture for input to the intelligence and command decision processes. Survival during conflict favors the side with the knowledge edge and superior technological capability. This article reflects on some contributions to the research and development of acoustic signal-processing methods that benefit warfighters of the submarine force, the land force, and the sea mine countermeasures force. Examples are provided of the application of the principles and practice of acoustic system science and engineering to provide the warfighter with enhanced situational awareness.

Acoustic systems are either *passive*, in that they exploit the acoustic noise radiated by a source (its so-called *sound signature*), or *active*, where they insonify the target and process the echo information.

Submarine Sonar

Optimal Beamforming

The digitization (i.e., creating digital versions of the analog outputs of sensors so that they can be used by a digital computing system) of Australia's submarines occurred 35 years ago with the Royal Australian Navy Research Laboratory undertaking the research, development, and at-sea demonstration of advanced next-generation passive sonar signal-processing methods and systems to improve the reach of the sensors and to enhance the situational awareness of a submarine.

A passive sonar on a submarine consists of an array of hydrophones (either hull mounted or towed) that samples the underwater acoustic pressure field in both space and time. The outputs of the spatially distributed sensors are combined by a beamformer, so that signals from a chosen direction are coherently added while the effects of noise and interference from other directions are reduced by destructive interference. The beamformer appropriately weights the sensor outputs before summation so as to enhance the detection and estimation performance of the passive sonar system by improving the output signal-to-noise ratio. This improvement in the signal-to-noise ratio relative to that of a single sensor is referred to as the *array gain* (see Wage, 2018; Zurk, 2018).

After transformation from the time domain to the frequency domain, the hydrophone outputs are beamformed in the spatial frequency domain to produce a frequency-wave number power spectrum. (The wave number is the number of

wavelengths per unit distance in the direction of propagation.) A conventional delay-and-sum beamformer, where the weights are set to unity, is optimal in the sense that the output signal-to-noise ratio is a maximum for an *incoherent* noise field. However, when the noise field includes *coherent* sources (such as interference), then an adaptive beamformer that is able to maximize the output signal-to-noise ratio by applying a set of weights that are *complex numbers* is implemented. This has the effect of steering a null in the direction of any unwanted interference (a jammer).

Figure 1 shows a comparison of the frequency-wave number spectrum for an actual underwater acoustic field sensed by an array using the conventional weight vector (*left*) and an adaptive weight vector (*right*). The adaptive beamformer suppresses the side lobes and resolves the various contributions to the acoustic pressure field, which are shown as surfaces (ridges) associated with towed array self-noise (structural waves that propagate along the array in both axial directions, i.e., aft and forward); tow-vessel radiated noise observed at, and near, the forward end-fire direction (i.e., the direction of the longitudinal axis of the array) for the respective direct propagation path and the indirect (surface-reflected) multipath; and three surface ship contacts. The adaptive beamformer better delineates the various signal and noise sources that compose this underwater sound field (Ferguson, 1998).

Towed-Array Shape Estimation

When deep, submarines rely exclusively on their passive sonar systems to sense the underwater sound field for radiated noise from ships underway and antisubmarine active sonar transmissions. The long-range *search sonar* on a submarine consists of a thin flexible neutrally buoyant streamer fitted with a line array of hydrophones, which is towed behind the submarine. Towed arrays overcome two problems that limit the performance of hull-mounted arrays: the noise of the submarine picked up by the sensors mounted on the hull and the size of the acoustic aperture being constrained by the limited length of the submarine.

Unfortunately, submarines cannot travel in a straight line forever to keep the towed array straight, so the submarine is “deaf” when it undertakes a maneuver to solve the left-right ambiguity problem or to estimate the range of a contact by triangulation. Once the array is no longer straight but bowed, sonar contact is lost (**Figure 2**).

Rather than instrumenting the length of the array with heading sensors (compasses) to solve this problem, the idea was

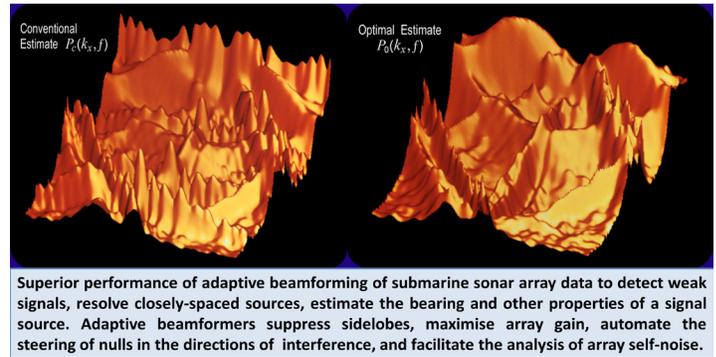


Figure 1. Left: estimated frequency-wave number power spectrum for a line array of hydrophones using the conventional frequency-domain beamforming method. The maximum frequency corresponds to twice the design frequency of the array. **Right:** similar to the left hand side but for an adaptive beamformer. From Ferguson (1998).

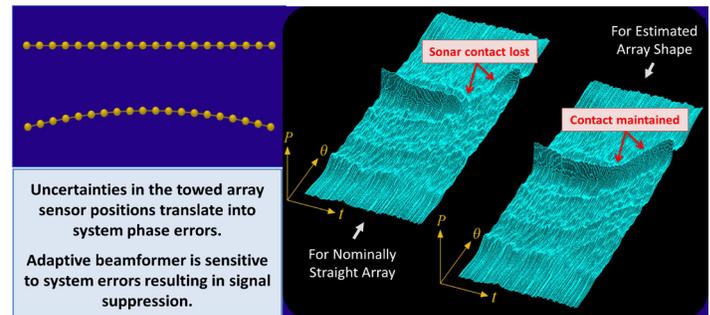


Figure 2. Left: a towed array is straight before the submarine maneuver, but it is bowed during the maneuver. **Right:** variation with bearing and time of the output of the beamformer before, during, and after the change of heading of the submarine. The total observation period is 20 minutes, and the data are real. When the array is straight, the contact appears on one bearing before the maneuver and on another bearing after the maneuver when the submarine is on its new heading and the array becomes straight again. Estimating the array shape (or the coordinates of the sensor positions in two dimensions) during the heading changes enables contact to be maintained throughout the submarine maneuver. From Ferguson (1993a).

to reprocess the hydrophone data so that the shape of the array could be estimated at each instant during a submarine maneuver when the submarine changes course so that it can head in another direction (Ferguson, 1990). The estimated shape had to be right because a nonconventional (or adaptive) beamformer was used to process the at-sea data. Uncertain knowledge of the sensor positions results in the signal being suppressed and the contact being lost. The outcome is that submariners maintain their situational awareness at all times, even during turns.

Compiling the Air Picture While at Depth

During World War II, aircraft equipped with centimeter-wavelength radars accounted for the bulk of Allied defeats of U-boats from 1943 to 1945. U-boats spent most of their time surfaced, running on diesel engines and diving only when attacked or for rare daytime torpedo attacks (Lansford and Tucker, 2012).

In 1987, a series of at-sea experiments using Australian submarines and maritime patrol aircraft demonstrated the detection, classification, localization, and tracking of aircraft using a towed array deployed from a submarine (Ferguson and Speechley, 1989). The results subsequently informed the full-scale engineering development of the Automated Threat Overflight Monitoring System (ATOMS) for the US Navy submarine force. ATOMS offers early warning/long-range detection of threat aircraft by submerged submarines via towed arrays (Friedman, 2006).

In a seminal paper, Urick (1972) indicated the possible existence of up to four separate contributions to the underwater sound field created by the presence of an airborne acoustic source. **Figure 3, left**, depicts each of these contributions: direct refraction, one or more bottom reflections, the evanescent wave, and sound scattered from a rough sea surface. When the aircraft flies overhead, its radiated acoustic noise travels via the direct refraction path where it is received by a hydrophone (after transmission across the air-sea interface). Now, the ratio of the speed of sound in air to that in water is 0.22; at a critical angle of incidence (θ_c) = 13°, the **black arrows** in **Figure 3** (the incident ray in air and the refracted ray along the sea surface) depict the propagation path of a critical ray. The transmission of aircraft noise across the air-sea interface occurs when the angle of incidence is less than θ_c (**Figure 3, red arrows**). For angles of incidence greater than θ_c , the radiated noise of the aircraft is reflected from the sea surface, with no energy being transmitted across the air-sea interface (**Figure 3, blue arrows**).

The reception of noise from an aircraft via the direct path relies on the aircraft being overhead. This transitory phenomenon lasts for a couple of seconds; its intensity and duration depend on the altitude of the aircraft and the depth of the receiver. Submariners refer to the overhead transit as an aircraft “on top.” Urick (1972) estimated the altitude of the aircraft by measuring the intensity and duration of the acoustic footprint of the aircraft. An alternative approach is to measure the variation in time of the instantaneous frequency corresponding to the propeller blade rate of the

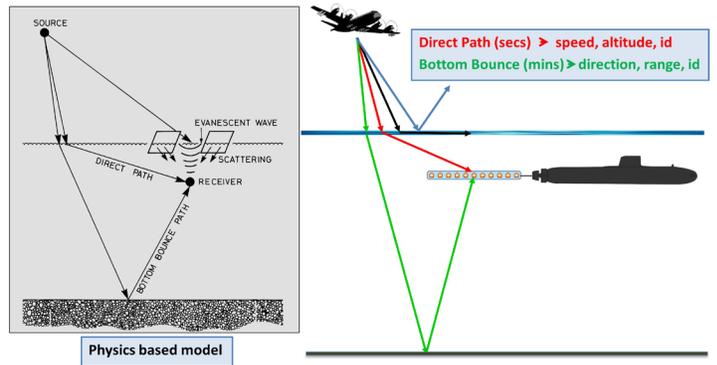


Figure 3. Left: contributions made by an acoustic source in the air to the sound field at a receiver in the sea. From Urick (1972). **Right:** ray paths for a bottom bounce (**green arrows**), direct refraction (**red arrows**), critical angle where the refracted ray lies along the sea surface (**black arrows**), and sea surface reflection (**blue arrows**). See text for further explanation.

aircraft and then extract tactical information on the speed (using the Doppler effect), altitude (using the rate of change of the instantaneous frequency), and identification (from the source/rest frequency of the blade rate) of the aircraft (Ferguson, 1996). For a submariner, the upside of an aircraft on top is identification of the aircraft and its mission profile by processing the output of a single hydrophone without giving away the position of the submarine; the downside is that there is no early warning.

The long-range detection of a submarine relies on reception of the radiated noise from the aircraft after undergoing one (or more) reflections from the seafloor (Ferguson and Speechley, 2009). The intensity of the noise from the aircraft received via direct refraction is considerably stronger (by 20 to 30 dB) than for propagation involving a seafloor reflection, so a towed array is required to detect the Doppler-shifted propeller blade rate (and its harmonics) and to measure its angle of arrival (bearing). A submarine with its towed array deployed has many minutes warning of an approaching aircraft. In **Figure 3, right**, the **green arrows** depict a bottom bounce path that enables the long-range detection of an aircraft.

Estimating the Instantaneous Range of a Contact

The towed-array sonar from a submarine is only able to measure the bearing of a contact. To estimate its range, the submarine must undertake a maneuver to get another fix on the contact and then triangulate its position. This process takes tens of minutes. Another approach is to use wide-aperture array sonar, which exploits the principle of passive ranging by wave front curvature to estimate the bearing and range of the contact at any instant, without having to perform a submarine maneuver (see **Figure 4, left**). The radius of curvature of the wave front equates to the range. Measurement of the differences in the arrival times (or time delays τ_{12} and τ_{23}) of the wave front at two

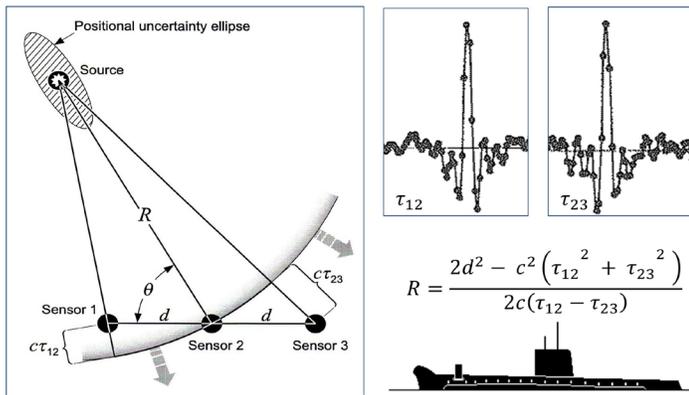


Figure 4. Left: schematic showing the source-sensor configuration for passive ranging by a wave front curvature. Right: cross-correlation functions for the outputs of sensors 1,2 and sensors 2,3. The time lags corresponding to the peaks in the two cross-correlograms provide estimates of the time delays $\tau_{1,2}$ and $\tau_{2,3}$. These time delays are substituted into the passive ranging equation to calculate the range (R). c , Speed of sound traveling in the underwater medium; d , intersensor separation distance. From Ferguson and Cleary (2001).

adjacent pairs of sensors enables estimation of the source range from the middle array and the source bearing with respect to the longitudinal axis of the wide-aperture array. Measuring a time delay involves cross-correlating the receiver outputs. The time delay corresponds to the time lag at which the cross-correlation function attains its maximum value.

Figure 4, right, shows the cross-correlation functions for sensor pairs 1,2 and 2,3. In practice, arrays of sensors (rather than the single sensors shown here) are used to provide array gain. It is the beamformed outputs that are cross-correlated to improve the estimates of the time delays. Each array samples the underwater sound field for 20 s, then the complex weights are calculated so that an adaptive beamformer maximizes the array gain, suppresses side lobes, and automatically steers nulls in the directions of interference. The process is repeated every 20 s because the relative contributions of the signal, noise, and interference components to the underwater sound field can vary over a period of minutes. In summary, beamforming and prefiltering suppress extraneous peaks, which serve to highlight the peak associated with a contact (Ferguson, 1993b).

Battlefield Acoustics

Extracting Tactical Information with a Microphone

In 1995, sonar system research support to the Royal Australian Navy Oberon Submarine Squadron was concluded, which ushered in a new research and development program for the Australian Army on the use of acoustics on the battlefield. Battlefield acoustics had gone out of fashion and be-

came dormant after the invention of radar in the 1930s. The Australian Army's idea was that the signal-processing techniques developed for the hydrophones of a submarine be used for microphones deployed on the battlefield. The goal was "low-cost intelligent acoustic-sensing nodes operating on shoestring power budgets for years at a time in potentially hostile environments without hope of human intervention" (Hill et al., 2004).

The sensing of sound on the battlefield makes sense because acoustic sensors are passive;

- sound propagation is not limited by line of sight;
- the false-alarm rate is negligible due to smart signal processing;
- unattended operation with only minimal tactical information (what, when, where) being communicated to a central monitoring facility;
- sensors are light weight, low cost, compact, and robust;
- acoustic systems can cue other systems such as a cameras, radars, and weapons;
- acoustic signatures of air and ground vehicles enable rapid classification of type of air or ground vehicle as well as weapon fire; and
- military activities are inherently noisy.

The submarine approach to wide-area surveillance requires arrays with large numbers of closely spaced sensors (submarines bristle with sensors) and the central processing of the acoustic sound field information on board the submarine. In contrast, wide-area surveillance of the battlefield is achieved by dispersing acoustic sensor nodes throughout the surveillance area, then networking them and using decentralized data fusion to compile the situational awareness picture. On the battlefield, only a minimal number of sensors (often, just one) is required for an acoustic-sensing node to extract the tactical information (position, speed, range at the closest point of approach to the sensor) of a contact and to classify it. For example, in 2014, the Acoustical Society of America (ASA) Technical Committee on Signal Processing in Acoustics posed an international student challenge problem (available at bit.ly/1rjN3AG) where the students were given a 30-s sound file of a truck traveling past a microphone (Ferguson and Culver, 2014). By processing the sound file, the students were required to extract the tactical information so that they could respond to the following questions.

- What is the speedometer reading?
- What is the tachometer reading?
- How many cylinders does the engine have?

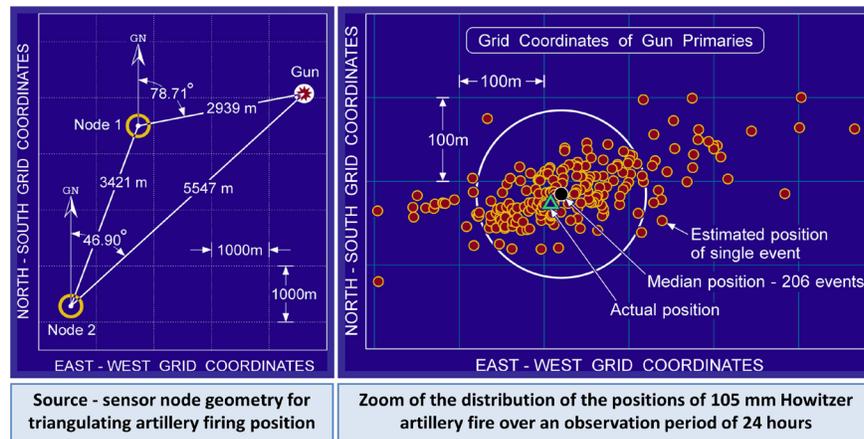


Figure 5. Left: source-sensor node geometry. Right: zoom acoustic locations of artillery fire. From Ferguson et al. (2002).

- When is the vehicle at the closest point of approach to the microphone?
- What is the range to the vehicle at the closest point of approach?

The solution to this particular problem, together with the general application of acoustic signal-processing techniques to extract tactical information using one, two, or three microphones can be found elsewhere (Ferguson, 2016). The speed of the truck is 20 km/h and the tachometer reading is 2,350 rpm. It has 6 cylinders and its closest point of approach to the microphone is 35 m.

Locating the Point of Origin of Artillery and Mortar Fire

Historically, sound ranging, or the passive acoustic localization of artillery fire, had its genesis in World War I (1914-1918). It was the procedure for locating the point where an artillery piece was fired by using calculations based on the relative time of arrival of the sound impulse at several accurately positioned microphones. Gun ranging fell into disuse, and it was replaced by radar that detected the projectile once it was fired. However, radars are active systems (which make them vulnerable to counter attack) and so the Army revisited sound ranging with a view to complement weapon-locating radar.

Figure 5, left, shows two acoustic nodes locating the point of origin of 206 rounds of 105 mm Howitzer fire by triangulation using angle-of-arrival measurements of the incident wave front at the nodes. Zooming in on the location of the firing point shows the scatter in the grid coordinates of the gun primaries due to the meteorological effects of wind and temperature variations on the propagation of sound in the atmosphere (Figure 5, right). Ferguson et al. (2002) showed the results of localizing indirect weapon fire using acoustic sensors in an extensive series of field experiments conducted during army field exercises over many years. This work in-

formed the full-scale engineering development and production of the Unattended Transient Acoustic Measurement and Signal Intelligence (MASINT) System (UTAMS) by the US Army Research Laboratory for deployment by the US Army during Operation Iraqi Freedom (2003-2011). This system had an immediate impact, effectively shutting down rogue mortar fire by insurgents (UTAMS, 2018).

Locating a Hostile Sniper's Firing Position

The firing of a sniper's weapon is accompanied by two acoustic transient events: a muzzle blast and a ballistic shock wave. The muzzle blast transient is generated by the discharge of the bullet from the firearm. The acoustic energy propagates at the speed of sound and expands as a spherical wave front centered on the point of fire.

The propagation of the muzzle blast is omnidirectional, so it can be heard from any direction including those directions pointing away from the direction of fire. If the listener is positioned in a direction *forward* of the firer, then the ballistic shock wave is heard as a loud sharp crack (or sonic boom) due to the supersonic speed of travel of the projectile along its trajectory. Unlike the muzzle blast wave front, the shock wave expands as a conical surface with the trajectory and nose of the bullet defining the axis and apex, respectively, of the cone (see Figure 6, top left). Also, the point of origin of the shock wave is the detach point on the trajectory of the bullet (see Figure 6, top right). In other words, with respect to the position of the receiver, the detach point is the position on the trajectory of the bullet from where the shock wave emanates. The shock wave arrives before the muzzle blast so, instinctively, a listener looks in the direction of propagation of the shock wave front, which is away from the direction of the shooter. It is the direction of the muzzle blast that coincides with the direction of the sniper's firing point.

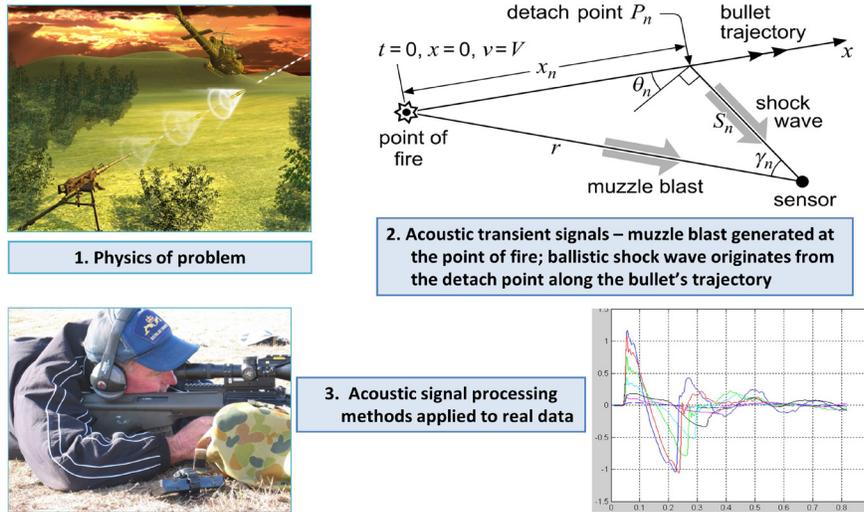


Figure 6. Top: left, image showing the conical wave front of the ballistic shock wave at various detach points along the trajectory of the supersonic bullet; right, source-sensor geometry and bullet trajectory. Bottom: left, source of small arms fire; right, “N”-shaped waveforms of ballistic shock waves for increasing miss distances. From Ferguson et al. (2007).

Detecting the muzzle blast signals and measuring the time difference of arrival (time delay) of the wave front at a pair of spatially separated sensors provides an estimate of the source direction. The addition of another sensor forms a wide aperture array configuration of three widely spaced collinear sensors. As discussed in *Estimating the Instantaneous Range of a Contact* for submarines, passive ranging by the wave front curvature involves measuring the time delays for the wave front to traverse two pairs of adjacent sensors to estimate the source range from the middle sensor and a source bearing with respect to the longitudinal axis of the three-element array. Also, by measuring the differences in both the times of arrival and the angles of arrival of the ballistic shock wave and the muzzle blast enables estimation of the range and bearing of the shooter (Lo and Ferguson, 2012). In the absence of a muzzle blast wave (due to the rifle being fitted with a muzzle blast suppressor or the excessive transmission loss of a long-range shot), the source (Figure 6, bottom left) can still be localized using only time delay measurements of the shock wave at spatially distributed sensor nodes (Lo and Ferguson, 2012). Finally, Ferguson et al. (2007) showed that the caliber of the bullet and its miss distance can be estimated using a wideband piezoelectric (quartz) dynamic pressure transducer by measuring the peak pressure amplitude and duration of the “N” wave (see Figure 6, bottom right).

High-Frequency Sonar Tomographic Imaging Sonar

The deployment of sea mines threatens the freedom of the seas and maritime trade. Sea mines are referred to as asymmetric threats because the cost of a mine is disproportionately small compared with the value of the target. Also, area denial

is achieved with an investment many times less than the cost of mine-clearing operations.

Once a mine like object (MLO) is detected by a high-frequency (~100 kHz) mine-hunting sonar, the next step is to identify it. The safest way to do this is to image the mine at a suitable standoff distance (say 250 m away). The acoustic image is required to reveal the shape and detail (features) of the object, which means the formation of a high-resolution acoustic image, with each pixel representing an area on the object ~1 cm long by ~1 cm wide. The advent of high-frequency 1-3 composite sonar transducers with bandwidths comparable to their center frequencies (i.e., $Q \approx 1$) means that the along-range resolution $\delta r \approx 1$ cm. However, real aperture-receiving arrays have coarse cross-range resolutions of 1-10 m, which are range dependent and prevent high-resolution acoustic imaging. The solution is to synthesize a virtual aperture so that the cross-range resolution matches the along-range resolution (Ferguson and Wyber, 2009). The idea of a tomographic imaging sonar is to circumnavigate the mine at a safe standoff distance while simultaneously insonifying the underwater scene, which includes the object of interest. Rather than viewing the scene many times from only one angle (which would improve detection but do nothing for multispect target classification), the plan is to insonify the object only once at a given angle but to repeat the process for many different angles. This idea is analogous to spotlight synthetic aperture radar.

Tomographic sonar image formation is based on image reconstruction from projections (Ferguson and Wyber, 2005). Populating the projection (or observation) space with measurement data requires insonifying the object to be imaged

from all possible directions and recording the impulse response of the backscattered signal (or echo) as a function of aspect angle (see **Figure 7, left**). Applying the inverse Radon transform method or two-dimensional Fourier transform reconstruction algorithm to the two-dimensional projection data enables an image to be formed that represents the two-dimensional spatial distribution of the acoustic reflectivity function of the object when projected on the imaging plane (see **Figure 7, bottom right**). The monostatic sonar had a center frequency of 150 kHz and a bandwidth of 100 kHz, so the range resolution of the sonar transmissions is less than 1 cm. About 1,000 views of the object were recorded so that the angular increment between projections was 0.35° . Distinct features in the measured projection data (see **Figure 7, left**) are the sinusoidal traces associated with point reflectors visible over a wide range of aspect angles (hence, the term “sinogram”). Because the object is a truncated cone, which is radially symmetric, the arrival time is constant for the small specular reflecting facets that form the outermost boundary (rim at the base) of the object. **Figure 7, top right**, shows a photograph of a truncated cone practice mine (1 m diameter base, 0.5 m high), which is of fiberglass construction with four lifting lugs and a metal end plate mounted on the top surface. **Figure 7, bottom right**, shows the projection of the geometrical shape of the object and acoustic highlights on the image plane. It shows the outer rim, four lifting lugs, and acoustic highlights associated with the end plate. Hence, tomographic sonar imaging is effective for identifying mines at safe standoff distances. Unlike real aperture sonars, the high resolution of the image is independent of the range.

Naval Base and Port Asset Protection

The suicide bombing attack of the USS COLE in October 2000 prompted the expansion of a program on the research and development of advanced mine-hunting sonars to include other asymmetric threats: fast inshore attack craft (FIAC), divers, and unmanned underwater vehicles. The detection, classification, localization, and tracking of these modern asymmetric threats were demonstrated using both passive and active sonar signal-processing techniques. The passive methods had already proven themselves in battlefield acoustic applications.

The rotating propeller of a FIAC generates a wake of bubbles that persists for minutes. When the wake is insonified by high-frequency active sonar transmissions, echoes are received from the entire wake, which traces out the trajectory of the watercraft. Insurgents rely on surprise and fast attack, so it is necessary to automate the detection, localization, and

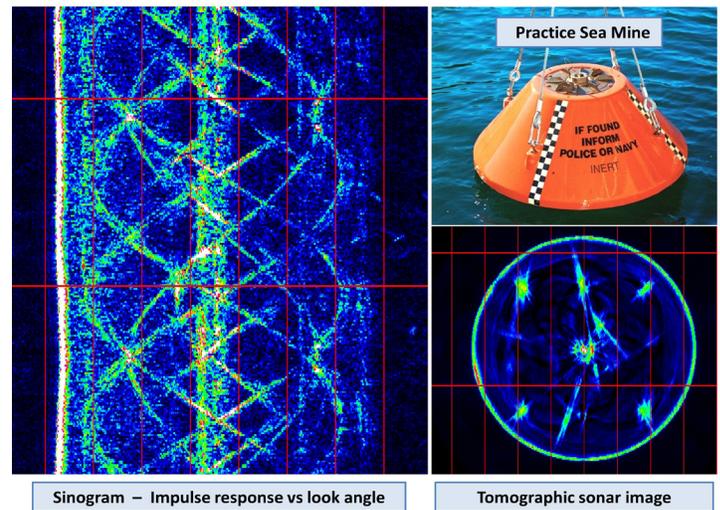


Figure 7. Left: two-dimensional projection data or intensity plot of the impulse response of the received sonar signal as a function of time (horizontal axis) and insonification angle (vertical axis). **Right: top,** photograph of the object; **bottom,** tomographic sonar image of the object. From Ferguson and Wyber (2005).

tracking processes because a FIAC attack can happen in as little as 20 s. A high-frequency high-resolution active sonar (or forward-looking mine-hunting sector scan sonar; see **Figure 8, top**) was adapted to *automate* the detection, localization, tracking, and classification of a fast inshore craft in a very shallow water (7 m deep), highly-cluttered environment. The capability of the system was demonstrated at the HMAS PENGUIN naval base in Middle Harbour, Sydney.

A sequence of sonar images for 100 consecutive pings (corresponding to an overall observation period of 200 s) captured the nighttime intrusion by a small high-speed surface craft. **Figure 8, bottom**, shows the sonar image for ping 61 during the U-turn of the craft. The wake of the craft is clearly observed in the sonar image. The clutter in the sonar display is bounded by the wake of the craft and is associated with the hulls of pleasure craft and the keels of moored yachts. The high-intensity vertical strip at a bearing of 6° is due to the cavitation noise generated by the rapidly rotating propeller of the craft. In this case, the receiver array and processor of the sonar act as a passive sonar with cavitation noise as the signal. This feature provides an immediate alert to the presence of the craft in the field of view of the sonar. The sonar echoes returned from the wake (bubbles) are processed to extract accurate range and bearing information to localize the source. The number of false alarms is reduced by range normalization and clutter map processing, which together with target position measurement, target detection/track initiation, and track maintenance are described elsewhere (Lo and Ferguson, 2004). For ping 61, the automated tracking

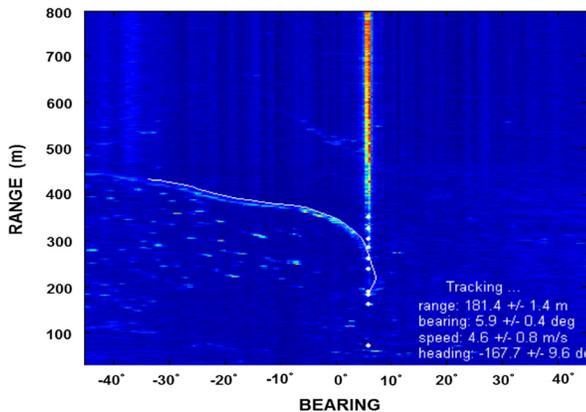
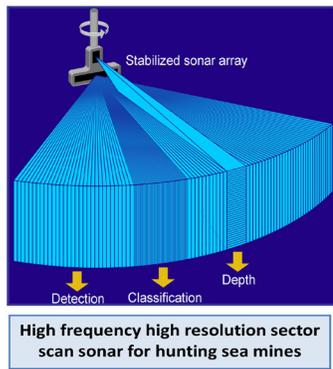


Figure 8. Top: Sector scan sonar showing the narrow receive beams for detection and even narrower ones for classification. **Bottom:** sonar image for ping 61 as the fast surface watercraft continues its U-turn. From Ferguson and Lo (2011).

processor estimated the instantaneous polar position of the craft (i.e., the end point of the wake) to be 181.4 m and 5.9°, the speed of the craft to be 4.6 m/s, and the heading to be -167.7°. Any offset of the wake from the track is caused by the wake drifting with the current.

The sonar is the primary sensor and, under the rules of engagement, a rapid layered response is now possible using a combination of nonlethal, less than lethal, and (last resort) lethal countermeasures.

Conclusion

The application of the principles and practice of acoustic systems science and engineering has improved the detection, classification, localization, and tracking processes for the Submarine Force, Land Force, and Mine Countermeasures Force, leading to enhanced situational awareness. Acoustic systems will continue to play a crucial role in operational systems, with new sensing technologies and signal-processing and data fusion methods being developed for the next generation of defense forces and homeland security.

Acknowledgments

I am grateful to the following Fellows of the Acoustical Society of America, who taught and inspired me over three decades: Ed Sullivan, Jim Candy, and Cliff Carter in sonar signal processing; Howard Schloemer in submarine sonar array design; Bill Carey, Doug Cato, and Gerald D'Spain in underwater acoustics; Tom Howarth and Kim Benjamin in ultrawideband sonar transducers; R. Lee Culver and Whitlow Au in high-frequency sonar; and Mike Scanlon in battle-field acoustics. In Australia, much was achieved through collaborative work programs with Ron Wyber in the research, development, and demonstration of acoustic systems science and engineering for defense.

References

- Ferguson, B. G. (1990). Sharpness applied to the adaptive beamforming of acoustic data from a towed array of unknown shape. *The Journal of the Acoustical Society of America* 88, 2695-2701.
- Ferguson, B. G. (1993a). Remedying the effects of array shape distortion on the spatial filtering of acoustic data from a line array of hydrophones. *IEEE Journal of Oceanic Engineering* 18, 565-571.
- Ferguson, B. G. (1993b). Improved time-delay estimates of underwater acoustic signals using beamforming and prefiltering techniques. In G. C. Carter (Ed.), *Coherence and Time Delay Estimation*. IEEE Press, New York, pp. 85-91.
- Ferguson, B. G. (1996). Time-frequency signal analysis of hydrophone data. *IEEE Journal of Oceanic Engineering* 21, 537-544.
- Ferguson, B. G. (1998). Minimum variance distortionless response beamforming of acoustic array data. *The Journal of the Acoustical Society of America* 104, 947-954.
- Ferguson, B. G. (2016). Source parameter estimation of aero-acoustic emitters using non-linear least squares and conventional methods. *IET Journal on Radar, Sonar & Navigation* 10(9), 1552-1560. <https://doi.org/10.1049/iet-rsn.2016.0147>.
- Ferguson, B. G., and Cleary, J. L. (2001). *In situ* source level and source position estimates of biological transient signals produced by snapping shrimp in an underwater environment. *The Journal of the Acoustical Society of America* 109, 3031-3037.
- Ferguson, B. G., Criswick, L. G., and Lo, K. W. (2002). Locating far-field impulsive sound sources in air by triangulation. *The Journal of the Acoustical Society of America* 111, 104-116.
- Ferguson, B. G., and Culver, R. L. (2014). International student challenge problem in acoustic signal processing. *Acoustics Today* 10(2), 26-29.
- Ferguson, B. G., and Lo, K. W. (2011). Sonar signal processing methods for detection and localization of fast surface watercraft and underwater swimmers in a harbor environment. *Proceedings of the International Conference on Underwater Acoustic Measurements*, Kos, Greece, June 20-24, 2011, pp. 339-346.
- Ferguson, B. G., Lo, K. W., and Wyber, R. J. (2007). Acoustic sensing of direct and indirect weapon fire. *Proceedings of the 3rd International Conference on Intelligent Sensors, Sensor Networks and Information Processing (ISSNIP 2007)*, Melbourne, Victoria, Australia, December 3-6, 2007, pp. 167-172.

Ferguson, B. G., and Speechley, G. C. (1989). Acoustic detection and localization of an ASW aircraft by a submarine. *The United States Navy Journal of Underwater Acoustics* 39, 25-41.

Ferguson, B. G., and Speechley, G. C. (2009). Acoustic detection and localization of a turboprop aircraft by an array of hydrophones towed below the sea surface. *IEEE Journal of Oceanic Engineering* 34, 75-82.

Ferguson, B. G., and Wyber, R. W. (2005). Application of acoustic reflection tomography to sonar imaging. *The Journal of the Acoustical Society of America* 117, 2915-2928.

Ferguson, B. G., and Wyber, R. W. (2009). Generalized framework for real aperture, synthetic aperture, and tomographic sonar imaging. *IEEE Journal of Oceanic Engineering* 34, 225-238.

Friedman, N. (2006). *World Naval Weapon Systems*, 5th ed. Naval Institute Press, Annapolis, MD, p. 659.

Hill, J., Horton, M., Kling, R., and Krishnamurthy, L. (2004). The platforms enabling wireless sensor networks. *Communications of the ACM* 47, 41-46.

Lansford, T., and Tucker, S. C. (2012). Anti-submarine warfare. In S. C. Tucker (Ed.), *World War II at Sea: An Encyclopedia*. ABC-CLIO, Santa Barbara, CA, vol. 1, pp. 43-50.

Lo, K. W., and Ferguson, B. G. (2004). Automatic detection and tracking of a small surface watercraft in shallow water using a high-frequency active sonar. *IEEE Transactions on Aerospace and Electronic Systems* 40, 1377-1388.

Lo, K. W., and Ferguson, B. G. (2012). Localization of small arms fire using acoustic measurements of muzzle blast and/or ballistic shock wave arrivals. *The Journal of the Acoustical Society of America* 132, 2997-3017.

Unattended Transient Acoustic Measurement and Signal Intelligence (MASINT) System (UTAMS). (2018). Geophysical MASINT. *Wikipedia*. Available at https://en.wikipedia.org/wiki/Geophysical_MASINT. Accessed October 30, 2018.

Urick, R. J. (1972). Noise signature of an aircraft in level flight over a hydrophone in the sea. *The Journal of the Acoustical Society of America* 52, 993-999.

Wage, K. E. (2018). When two wrongs make a right: Combining aliased arrays to find sound sources. *Acoustics Today* 14(3), 48-56.

Zurk, L. M. (2018). Physics-based signal processing approaches for underwater acoustic sensing. *Acoustics Today* 14(3), 57-61.

BioSketch



Brian G. Ferguson is the Principal Scientist (Acoustic Systems), Department of Defence, Sydney, NSW, Australia. He is a fellow of the Acoustical Society of America, a fellow of the Institution of Engineers Australia, and a chartered professional engineer. In 2015, he was awarded the Acoustical Society of America Silver Medal for Signal Processing in Acoustics. In 2016, he received the Defence Minister's Award for Achievement in Defence Science, which is the Australian Department of Defence's highest honor for a defense scientist. More recently, he received the David Robinson Special Award in 2017 from Engineers Australia (College of Information, Telecommunications and Electronics Engineering).

Where do you read your *Acoustics Today*?

Take a photo of yourself reading a copy of **Acoustics Today** and share it on social media* (tagging #AcousticsToday, of course). We'll pick our favorite and include it in the next issue of **AT**. Don't do social media but still want to provide a picture? Email it to ksetzer@acousticalsociety.org.

* By submitting, the subject agrees to image being used in **AT** and/or on the **AT** website. Please do not include children in the image unless you can give explicit permission for use as their parent or guardian.

ASA President Lily Wang tells Captain America about all the amazing achievements women scientists have made in the field of acoustics. Photo taken by Subha Maruvada.



Snap, Crackle, and Pop: Theracoustic Cavitation

Michael D. Gray

Address:

Biomedical Ultrasonics,
Biotherapy and Biopharmaceuticals
Laboratory (BUBBL)
Institute of Biomedical Engineering
University of Oxford
Oxford OX3 7DQ
United Kingdom

Email:

michael.gray@eng.ox.ac.uk

Eleanor P. Stride

Address:

Biomedical Ultrasonics,
Biotherapy and Biopharmaceuticals
Laboratory (BUBBL)
Institute of Biomedical Engineering
University of Oxford
Oxford OX3 7DQ
United Kingdom

Email:

eleanor.stride@eng.ox.ac.uk

Constantin-C. Coussios

Address:

Biomedical Ultrasonics,
Biotherapy and Biopharmaceuticals
Laboratory (BUBBL)
Institute of Biomedical Engineering
University of Oxford
Oxford OX3 7DQ
United Kingdom

Email:

constantin.coussios@eng.ox.ac.uk

Emerging techniques for making, mapping, and using acoustically driven bubbles within the body enable a broad range of innovative therapeutic applications.

Introduction

The screams from a football stadium full of people barely produce enough sound energy to boil an egg (Dowling and Ffowcs Williams, 1983). This benign view of acoustics changes dramatically in the realm of focused ultrasound where biological tissue can be brought to a boil in mere milliseconds (ter Haar and Coussios, 2007). Although the millimeter-length scales over which these effects can act may seem “surgical,” therapeutic ultrasound (0.5-3.0 MHz) is actually somewhat of a blunt instrument compared with drug molecules (<0.0002 mm) and the cells (<0.1 mm) that they are intended to treat.

Is therapeutic acoustics necessarily that limiting? Not quite. Another key phenomenon, acoustic cavitation, has the potential to enable subwavelength therapy. Defined as the linear or nonlinear oscillation of a gas or vapor cavity (or “bubble”) under the effect of an acoustic field, cavitation can enable preferential absorption of acoustic energy and highly efficient momentum transfer over length scales dictated not only by the ultrasound wavelength but also by the typically micron-sized diameter of therapeutic bubbles (Coussios and Roy, 2008). Under acoustic excitation, such bubbles act as “energy transformers,” facilitating conversion of the incident field’s longitudinal wave energy into locally enhanced heat and fluid motion. The broad range of thermal, mechanical, and biochemical effects (“theracoustics”) resulting from ultrasound-driven bubbles can enable successful drug delivery to the interior of cells, across the skin, or to otherwise inaccessible tumors; noninvasive surgery to destroy, remove, or debulk tissue without incision; and pharmacological or biophysical modulation of the brain and nervous system to treat diseases such as Parkinson’s or Alzheimer’s.

Where do these bubbles come from in the human body? Given that nucleating (forming) a bubble within a pure liquid requires prohibitively high and potentially unsafe acoustic pressures (10-100 atmospheres in the low-megahertz frequency range), cavitation has traditionally been facilitated by injection of micron-sized bubbles into the bloodstream. However, the currently evolving generation of therapeutic applications requires the development of biocompatible submicron cavitation nucleation agents that are of comparable size to both the biological barriers they need to cross and the size of the drugs alongside which they frequently travel. Furthermore, the harnessing and safe application of the bioeffects brought about by ultrasonically driven bubbles requires the development of new techniques capable of localizing and tracking cavitation in real time at depth within the human body. And thus begins our journey on making, mapping, and using bubbles for “theracoustic” cavitation.

Making Bubbles

An ultrasound field of sufficient intensity can produce bubbles through either mechanical or thermal mechanisms or a combination of the two. However, the energy that is theoretically required to produce an empty cavity in pure water is significantly higher than that needed to generate bubbles using ultrasound in practice. This discrepancy has been explained by the presence of discontinuities, or nuclei, that provide preexisting surfaces from which bubbles can evolve. The exact nature of these nuclei in biological tissues is still disputed, but both microscopic crevices on the surfaces of tissue structures and submicron surfactant-stabilized gas bubbles have been identified as likely candidates (Fox and Hertzfeld, 1954; Atchley and Prosperetti, 1989).

The energy required to produce cavitation from these so-called endogenous nuclei is still comparatively large, and so, for safety reasons, it is highly desirable to be able to induce reproducible cavitation activity in the body using ultrasound frequencies and amplitudes that are unlikely to cause significant tissue damage. Multiple types of synthetic or exogenous nuclei have been explored to this end. To date, the majority of studies have employed coated gas microbubbles widely used as contrast agents for diagnostic imaging (see article in *Acoustics Today* by Matula and Chen, 2013). A primary disadvantage of microbubbles is that their size (1-5 μm) prevents their passing out of blood vessels (extravasating) into the surrounding tissue. As a consequence, cavitation is restricted to the blood stream. The bubbles are also relatively unstable, having a half-life of about 2 minutes once injected.

There has consequently been considerable research into alternative nuclei. These include solid nanoparticles with hydrophobic cavities that can act as artificial crevices (Rapoport et al., 2007; Kwan et al., 2015). Such particles have much longer circulation times (tens of minutes) and are small enough to diffuse out of the bloodstream into the surrounding tissue. Nanoscale droplets of volatile liquids, such as perfluorocarbons, have also been investigated as cavitation nuclei (see *Acoustics Today* article by Burgess and Porter, 2015). These are similarly small enough to circulate in the bloodstream for tens of minutes and to extravasate. On exposure to ultrasound, the liquid droplet is vaporized to form a microbubble. A further advantage of artificial cavitation nuclei is that they can be used as a means of encapsulating therapeutic material to provide spatially targeted delivery. This can significantly reduce the risk of harmful side effects from highly toxic chemotherapy drugs.

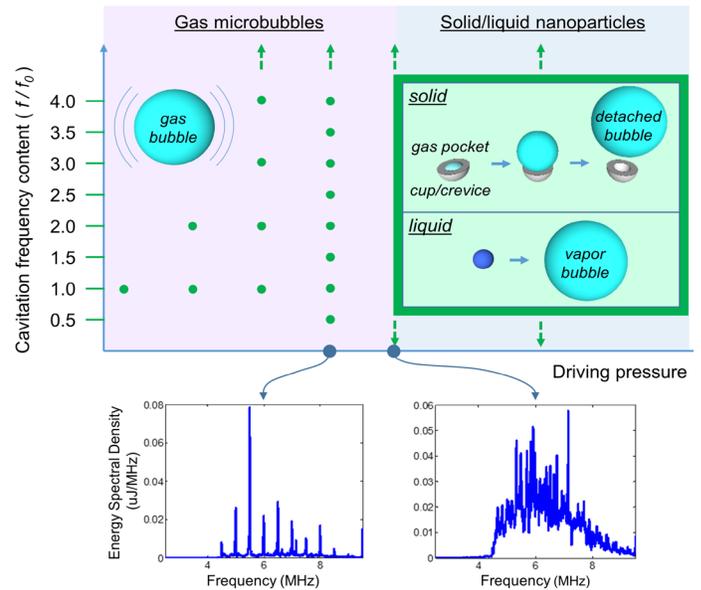


Figure 1. Illustrations of bubble emission behavior as a function of driving pressure. At moderate pressures, harmonics (integer multiples of the driving frequency [f_0]) are produced, followed by fractional harmonics at higher pressures and eventually broadband noise (dashed lines indicate that the frequency range extends beyond the scale shown). Microbubbles (top left) will generate acoustic emissions even at very low pressures, whereas solid and liquid nuclei (top right) require significant energy input to activate them and so typically produce broadband noise. The representative frequency spectra showing harmonic (bottom left) and broadband (bottom right) components were generated from cavitation measurements with $f_0 = 0.5$ MHz.

Cavitation agents also have an advantage over many other drug delivery devices because their acoustic emissions can be detected from outside the body, enabling both their location and dynamic behavior to be tracked. Even at relatively low-pressure amplitudes such as those used in routine diagnostic ultrasound, microbubbles respond in a highly nonlinear fashion and hence their emissions contain harmonics of the driving frequency (Arvanitis et al., 2011). As the pressure increases so does the range of frequencies in the emission spectrum, which will include fractional harmonics and eventually broadband noise (Figure 1). More intense activity, which is normally associated with more significant biological effects, will produce broadband noise. Liquid and solid cavitation nuclei require activation pressures that are above the threshold for violent (inertial) bubble collapse and thus these agents always produce broadband emissions.

Mapping Bubbles

In both preclinical in vivo and clinical studies, a broad spectrum of bubble-mediated therapeutic effects has been demonstrated, ranging from the mild (drug release and intracellular delivery, site-specific brain stimulation) to the malevolent (volumetric tissue destruction). This sizable

range of possible effects and the typical clinical requirement for only a specific subset of these at any one instance underscores the need for careful monitoring during treatment. Despite the myriad bubble behaviors and resulting bioeffects, very few are noninvasively detectible in vivo. For example, light generation during inertial cavitation (sonoluminescence) produced under controlled laboratory conditions may be detectible one meter away, but in soft tissue, both the relatively weak light production and its rapid absorption renders in vivo measurement effectively impossible. Magnetic resonance (MR) techniques, known for generating three-dimensional anatomic images, may also be used to measure temperature elevation, with clinically achievable resolution on the order of 1°C, 1 s, and 1 mm (Rieke and Pauly, 2008). However, because these techniques are generally agnostic to the cause(s) of heating, they cannot mechanistically identify bubble contributions to temperature elevation nor can they generally indicate nonthermal actions of bubble activity. On the basis of high-resolution availability of bubble-specific response cues, active and passive ultrasonic methods appear best suited for noninvasive clinical monitoring.

Active Ultrasound

The enhanced acoustic-scattering strength of a bubble excited near resonance (Ainslie and Leighton, 2011) is exploitable with diagnostic systems that emit and receive ultrasound pulses; bubbles can be identified as regions with an elevated backscatter intensity relative the low-contrast scattering that is characteristic of soft tissues and biological liquids. Intravenous introduction of microbubbles may therefore substantially improve the diagnostic visibility of blood vessels (where microbubbles are typically confined) by increasing both echo amplitude and bandwidth (Stride and Coussios, 2010). Spatial resolution approaching 10 μm may be clinically feasible with “super-resolution” methods employing microbubbles exposed to a high-frame rate sequence of low-amplitude sound exposures (Couture et al., 2018), yielding in vivo microvascular images with a level of detail that was previously unthinkable with diagnostic ultrasound.

Active ultrasound transmissions temporally interleaved with therapeutic ultrasound pulses have been used for bubble detection and tracking (Li et al., 2014). This timing constraint means that cavitation activity occurring during therapeutic ultrasound exposures (often thousands of cycles) would be missed, and with it, information about the therapy process would remain unknown. This limitation is especially important if using solid cavitation nuclei, which are essentially anechoic unless driven with a low-frequency excitation (e.g., therapy beam).

Passive Ultrasound

Passive acoustic mapping (PAM) methods are intended to detect, localize, and quantify cavitation activity based on analysis of bubble-scattered sound (for a review of the current art, see Haworth et al., 2017; Gray and Coussios, 2018). Unlike active-imaging modalities, PAM images may be computed at any time during a therapeutic ultrasound exposure and are decoupled from restrictions on timing, pulse length, or bandwidth imposed by the means of cavitation generation. Because of this timing independence, PAM methods yield fundamentally different (and arguably more relevant) information about microbubble activity. For example, features in the passively received spectrum such as half-integer harmonics (**Figure 1**) may be used to distinguish cavitation types from each other and from nonlinearities in the system being monitored and therefore may indicate the local therapeutic effects being applied.

The processes for PAM image formation are illustrated in **Figure 2**. Suppose a single bubble located at position x_s radiates sound within a region of interest (ROI), such as a tumor undergoing therapeutic ultrasound exposure. Under ideal conditions, the bubble-emitted field propagates spherically, where it may be detected by an array of receivers, commonly a conventional handheld diagnostic array placed on the skin. PAM methods estimate the location of the bubble based on relative delays between array elements due to the curvature of the received wave fronts. This stands in contrast to conventional active methods that localize targets using the time delay between active transmission and echo reception.

After filtering raw array data to remove unwanted signals (such as the fundamental frequency of the active ultrasound that created the cavitation), PAM algorithms essentially run a series of spatial and temporal signal similarity tests consisting of two basic steps. First, the array data are steered to a point (x) in the ROI by adding time shifts to compensate for the path length between x and each array element. If the source actually was located at the steering location ($x = x_s$), then all the array signals would be temporally aligned. Second, the steered data are combined, which in its simplest form involves summation. In more sophisticated forms, the array signal covariance matrix is scaled by weight factors optimized to reduce interference from neighboring bubbles (Coviello et al., 2015).

Regardless of the procedural details, the calculation typically yields an array-average signal power for each steering location in the ROI. This quantity is maximized when the array has been steered to the source location, and the map has its best spatial resolution when interference from other sources has been suppressed. The processing illustration in **Figure 2** is in the time

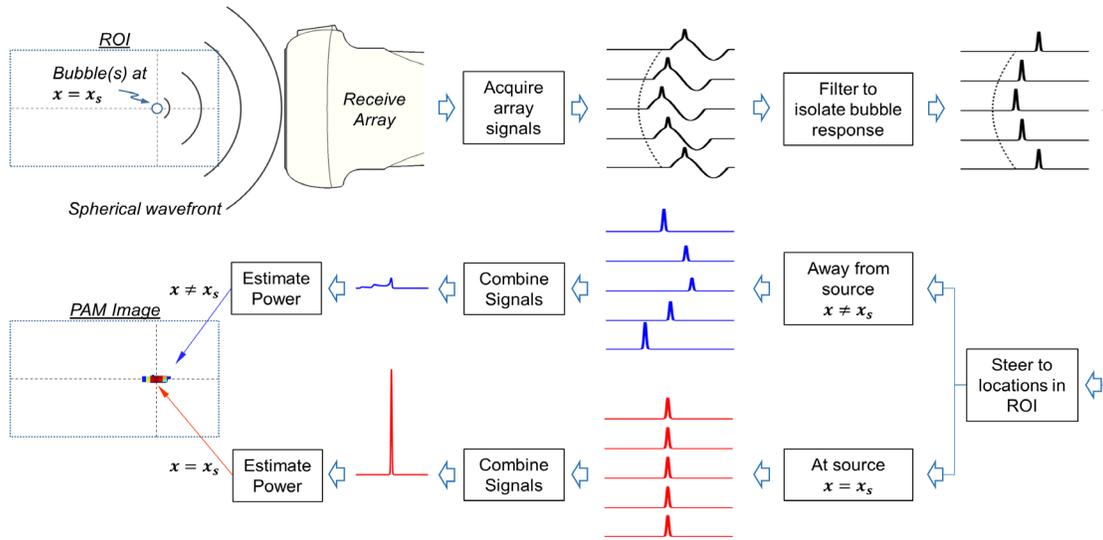


Figure 2. Time domain illustration of passive acoustic mapping (PAM) processing. Bubble emissions are received on an array of sensors. Signals (**black**) have relative delays that are characteristic of the distance between the array and the source location. After filtering the raw data to isolate either broadband or narrowband acoustic emissions of interest, the first processing step is to steer the array to a point in the region of interest (ROI) by applying time shifts to each array element. If the steered location matches that of the source ($x = x_s$), the signals will be time aligned (**red**); otherwise, the signals will be temporally misaligned (**blue**; $x \neq x_s$). The second processing step combines the time-shifted signals to estimate signal power; poorly correlated data will lead to a low power estimate while well-correlated data will identify the source. Repeating this process over a grid in the ROI leads to an image of source (bubble) strength (**bottom left**). The image has a dynamic range of 100, and **red** indicates a maximum value.

domain, but frequency domain implementations offer equivalent performance with potentially lower calculation times.

The roots of PAM techniques are found in passive beamforming research performed in the context of seismology, underwater acoustics, and radar. The utility of these techniques comes from their specific benefits when applied to noninvasive cavitation monitoring.

- Images are formed in the near field of the receive array so that sources may be identified in at least two dimensions (e.g., distance and angle).
- Received data may be filtered to identify bubble-specific emissions (half-integer harmonics or broadband noise elevation), thereby decluttering the image of nonlinear background scattering and identifying imaging regions that have different cavitation behaviors.
- A single diagnostic ultrasound array system can be used to provide both tissue imaging and cavitation mapping capabilities that are naturally coaligned so that the monitoring process can describe the tissue and bubble status before, during, and after therapy.
- Real-time PAM may allow automated control of the therapy process to ensure procedural safety.

For both passive and active ultrasonic methods, image quality and quantitative accuracy may be limited by uncertainties

in tissue sound speed and attenuation. Unlike MR or active ultrasonic methods, PAM data must be superimposed on tissue morphology images produced by other imaging methods to provide a context for treatment guidance and monitoring.

Clinical PAM Example

Over the last decade, PAM research has progressed from small-rodent to large-primate in vivo models, and recently, a clinical cavitation mapping dataset was collected during a Phase 1 trial of ultrasound-mediated liver heating for drug release from thermally sensitive liposomes (Lyon et al., 2018). **Figure 3** shows an axial computed tomography (CT) image of one trial participant, including the targeted liver tumor. The incident therapeutic-focused ultrasound beam (FUS; **Figure 3, red arrow**) was provided by a clinically approved system, while the PAM data (**Figure 3, blue arrow**) was collected using a commercially available curvilinear diagnostic array. The CT-overlaid PAM image (and movie showing six consecutive PAM frames, see acousticstoday.org/gray-media) was generated using a patient-specific sound speed estimate and an adaptive beamformer to minimize beamwidth. Although the monitoring was performed over an hour-long treatment, only a small handful of cavitation events was detected (<0.1% of exposures). This was as expected given that no exogenous microbubbles were used and the treatment settings

were intended to avoid thermally significant cavitation. This example shows the potential for real-time mapping of bubble activity in clinically relevant targets using PAM techniques.

Using Bubbles

Bubble Behaviors

The seemingly simple system of an ultrasonically driven bubble in a homogeneous liquid can exhibit a broad range of behaviors (Figure 4). In an unbounded medium, bubble wall motion induces fluid flow, sound radiation, and heating (Leighton, 1994) and may spur its own growth by rectified diffusion, a net influx of mass during ultrasonic rarefaction cycles. When a bubble grows sufficiently large during the rarefactional half cycle of the acoustic wave, it will be unable to resist the inertia of the surrounding liquid during the compressional half cycle and will collapse to a fraction of its original size. The resulting short-lived concentration of energy and mass further enhances sound radiation, heating rate, fluid flow, and particulate transport and can lead to a chemical reaction (sonochemistry) and light emission (sonoluminescence). Regarding the spatially and temporally intense action of this “inertial cavitation,” it has been duly noted that in a simple laboratory experiment “...one can create the temperature of the sun’s surface, the pressure of deep oceanic trenches, and the cooling rate of molten metal splatted onto a liquid-helium-cooled surface!” (Suslick, 1990, p. 1439).

Further complexity is introduced when the bubble vibrates near an acoustic impedance contrast boundary such as a glass slide in an *in vitro* experiment or blood vessel wall in tissue. Nonlinearly generated circulatory fluid flow known as “microstreaming” is produced as the bubble oscillates about its translating center of mass (Marmottant and Hilgenfeldt, 2003) and can both enhance transport of nearby therapeutic particles (drugs or sub-micron nuclei) and amplify fluid shear stresses that deform or rupture nearby cells (“microdeformation” in Figure 4).

Theracoustic Applications:

Furnace, Mixer, and Sniper Bubbles

Suitably nucleated, mapped, and controlled, all of the aforementioned phenomena find therapeutically beneficial applications within the human body. Here, we present a subset of the ever-expanding range of applications involving acoustic cavitation.

One of the earliest, and now most widespread, uses of therapeutic ultrasound is thermal, whereby an extracorporeal transducer is used to selectively heat and potentially destroy a well-defined tissue volume (“ablation”; Kennedy, 2005). A key challenge in selecting the optimal acoustic parameters to achieve this is the inevitable compromise between propaga-

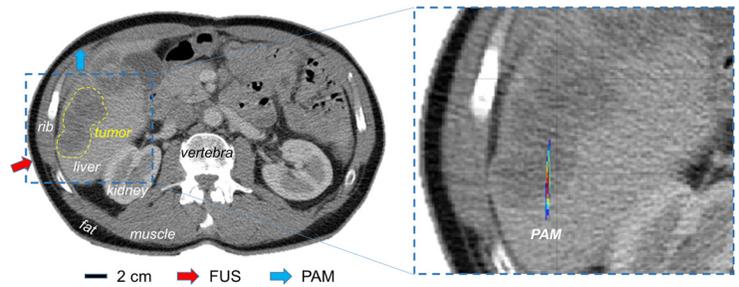


Figure 3. Example of passive cavitation mapping during a clinical therapeutic ultrasound procedure. **Left:** axial CT slice showing thoracic organs, including the tumor targeted for treatment, with red and blue arrows indicating the directions of therapeutic focused ultrasound (FUS) incidence and PAM data collection, respectively. **Right:** enlarged subregion (left, blue dashed-line box) in which a PAM image was generated. Maximum (red) and minimum (blue) color map intensities cover one order of magnitude. A video of six successive PAM frames spanning a total time period of 500 microseconds is available at (acousticstoday.org/gray-media).

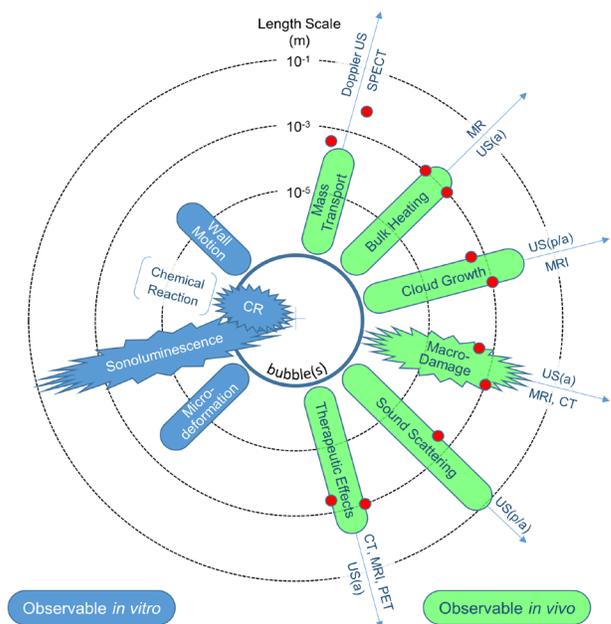


Figure 4. Illustration of bubble effects and length scales. **Green**, *in vivo* measurement feasibility; **jagged shapes** indicate reliance on inertial cavitation; **red dots**, best spatial resolution; **text around radial arrows**, demonstrated noninvasive observation methods. SPECT, single photon emission computed tomography; US, ultrasound; a, active; p, passive; MRI, magnetic resonance imaging; CT, X-ray computed tomography; PET, positron emission tomography.

tion depth, optimally achieved at lower frequencies, and the local rate of heating, which is maximized at higher frequencies. “Furnace” bubbles provide a unique way of overcoming this limitation (Holt and Roy, 2001); by redistributing part of the incident energy into broadband acoustic emissions that are more readily absorbed, inertial cavitation facilitates highly localized heating from a deeply propagating low-frequency wave (Coussios et al., 2007).

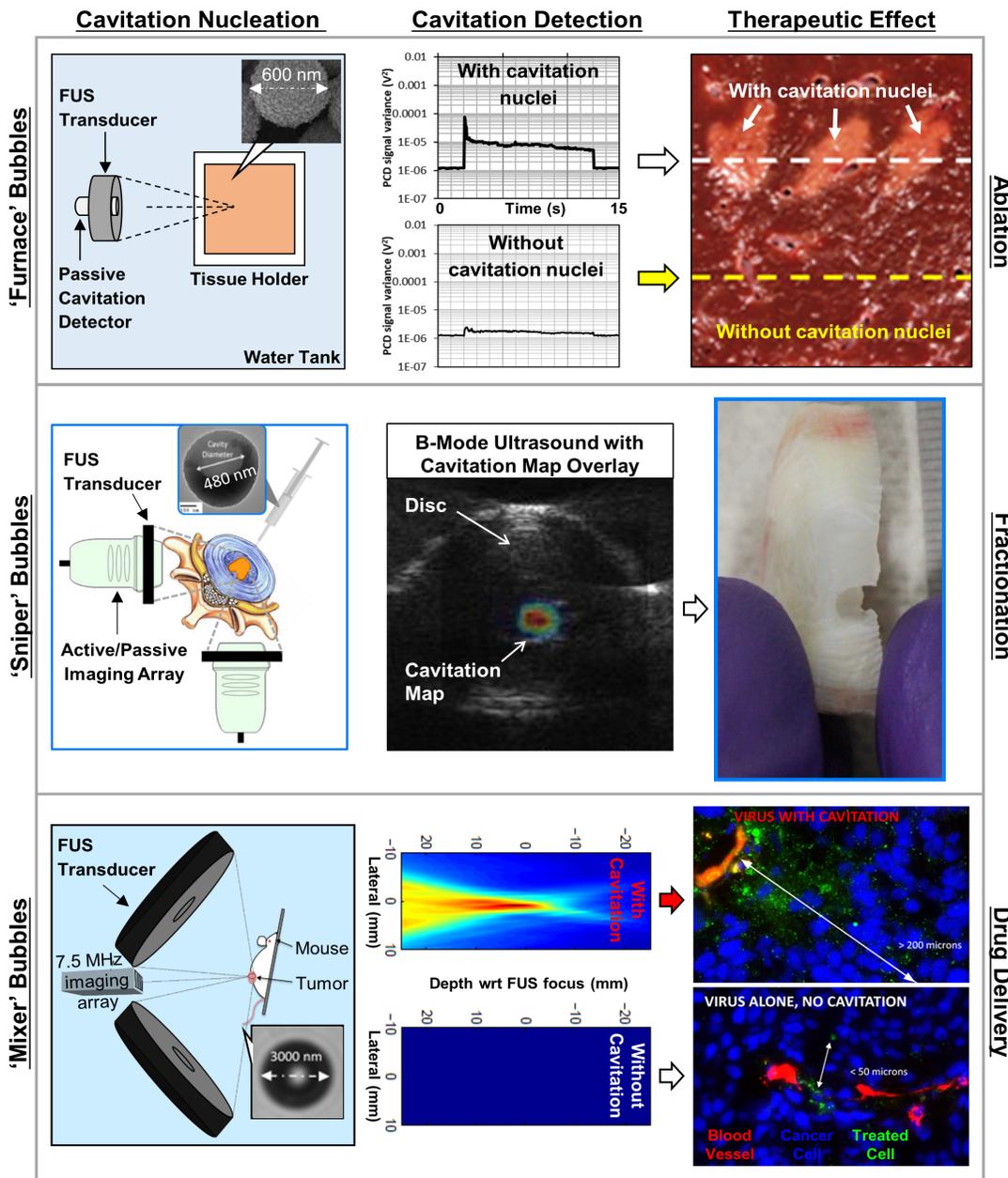


Figure 5. Nucleation and cavitation detection strategies for a range of emerging theracoustic applications. **Top row:** passively monitored cavitation-mediated thermal ablation nucleated by locally injected gas-stabilizing particles in liver tissue. **Center row:** dual-array passively mapped cavitation-mediated fractionation of the intervertebral disc, nucleated by gas-stabilizing solid polymeric particles. **Bottom row:** single-array passively mapped cavitation-enhanced drug delivery to tumors, following systemic administration of lipid-shelled microbubbles and an oncolytic virus. See text for a fuller explanation.

This process is illustrated in **Figure 5, top row**, using a sample of bovine liver exposed to FUS while monitoring the process with a single-element passive-cavitation detector (PCD). In the absence of cavitation nuclei, the PCD signal is weak and the tissue is unaltered. However, when using these same FUS settings in the presence of locally injected cavitation nuclei, the PCD signal is elevated by an order of magnitude and distinct regions of tissue ablation (in **Figure 5, top right, light pink regions**) correspond to exposures at three locations) are observed (Hockham, 2013). Because the ultrasound-based

monitoring of temperature remains extremely challenging, exploiting acoustic cavitation to mediate tissue heating enables the use of passive cavitation detection and mapping as a means of both monitoring (Jensen et al., 2013) and controlling (Hockham et al., 2010) the treatment in real time.

There are several emerging biomedical applications where the use of ultrasound-mediated heating is not appropriate due to the potential for damage to adjacent structures, and tissue disruption must be achieved by mechanical means

alone. In this context, “sniper”-collapsing bubbles come to the rescue by producing fluid jets and shear stresses that kill cells or liquify extended tissue volumes (Khokhlova et al., 2015). More recent approaches, such as in **Figure 5, center row**, have utilized gas-stabilizing solid nanoparticles to promote and sustain inertial cavitation activity. In this example, a pair of FUS sources initiate cavitation in an intervertebral disc of the spinal column while a pair of conventional ultrasound arrays is used to produce conventional (“B-mode”) diagnostic and PAM images during treatment. The region of elevated cavitation activity (**Figure 5, center row, center, red dot** on the B-mode image) corresponds to sources of broadband emissions detected and localized by PAM and also identifies the location and size of destroyed tissue. Critically, this theracoustic configuration has enabled highly localized disintegration of collagenous tissue in the central part of the disc without affecting the outer part or the spinal canal, potentially enabling the development of a new minimally invasive treatment for lower back pain (Molinari, 2012).

Acoustic excitation is not always required to act as the primary means of altering biology but can also be deployed synergistically with a drug or other therapeutic agent to enhance its delivery and efficacy. In this context, “mixer” bubbles have a major role to play; by imposing shear stresses at tissue interfaces and by transferring momentum to the surrounding medium, they can both increase the permeability and convectively transport therapeutic agents across otherwise impenetrable biological interfaces.

One such barrier is presented by the vasculature feeding the brain, which, to prevent the transmission of infection, exhibits very limited permeability that hinders the delivery of drugs to the nervous system. However, noninertial cavitation may reversibly open this so-called blood-brain barrier (see article in *Acoustics Today* by Konofagou, 2017). A second such barrier is presented by the upper layer of the skin, which makes it challenging to transdermally deliver drugs and vaccines without a needle. Recent studies have indicated that the creation of a “patch” containing not only the drug or vaccine but also inertial cavitation nuclei (Kwan et al., 2015) can enable ultrasound to simultaneously permeabilize the skin and transport the therapeutic to hundreds of microns beneath the skin surface to enable needle-free immunization (Bhatnagar et al., 2016). Last but not least, perhaps the most formidable barrier to drug delivery is presented by tumors where the elevated internal pressure, sparse vascularity, and dense extracellular matrix hinder the ability of anticancer drugs to reach cells far removed from blood vessels. Sustained inertial cavita-

tion nucleated by either microbubbles (Carlisle et al., 2013) or submicron cavitation nuclei (Myers et al., 2016) has been shown to enable successful delivery of next-generation, larger anticancer therapeutics to reach each and every cell within the tumor, significantly enhancing their efficacy.

An example is shown in **Figure 5, bottom row**, where a pair of FUS sources was used for in vivo treatment of a mouse tumor using an oncolytic virus (140 nm) given intravenously. In the absence of microbubbles, the PAM image (**Figure 5, bottom row, center**) indicates no cavitation, and only the treated cells (**Figure 5, bottom row, center, green**) are those directly adjacent to the blood vessel (**Figure 5, bottom row, center, red**). However, when microbubbles were coadministered with the virus, the penetration and distribution of treatment were greatly enhanced, correlating with broadband acoustic emissions associated with inertial cavitation in the tumor. Excitingly, noninvasively mapping acoustic cavitation mediated by particles that are coadministered and similarly sized to the drug potentially makes it possible to monitor and confirm successful drug delivery to target tumors during treatment for the very first time.

Final Thoughts

Acoustic cavitation demonstrably enables therapeutic modulation of a number of otherwise inaccessible physiological barriers, including crossing the skin, delivering drugs to tumors, accessing the brain and central nervous system, and penetrating the cell. Much remains to be done, both in terms of understanding and optimizing the mechanisms by which oscillating bubbles mediate biological processes and in the development of advanced, indication-specific technologies for nucleating, promoting, imaging, and controlling cavitation activity in increasingly challenging anatomical locations. Suitably nucleated, mapped, and controlled, therapeutic cavitation enables acoustics to play a major role in shaping the future of precision medicine.

Acknowledgments

We gratefully acknowledge the continued support over 15 years from the United Kingdom Engineering and Physical Sciences Research Council (Awards EP/F011547/1, EP/L024012/1, EP/K021729/1, and EP/I021795/1) and the National Institute for Health Research (Oxford Biomedical Research Centre). Constantin-C. Coussios gratefully acknowledges support from the Acoustical Society of America under the 2002-2003 F. V. Hunt Postdoctoral Fellowship in Acoustics. Last but not least, we are hugely grateful to all the clinical and postdoctoral research fellows, graduate students, and collaborators who have

contributed to this area of research and to the broader therapeutic ultrasound and “theracoustic” cavitation community for its transformative ethos and collaborative spirit.

References

- Ainslie, M. A., and Leighton, T. G. (2011). Review of scattering and extinction cross-sections, damping factors, and resonance frequencies of a spherical gas bubble. *The Journal of the Acoustical Society of America* 130, 3184-3208. <https://doi.org/10.1121/1.3628321>.
- Arvanitis, C. D., Bazan-Peregrino, M., Rifai, B., Seymour, L. W., and Coussios, C. C. (2011). Cavitation-enhanced extravasation for drug delivery. *Ultrasound in Medicine & Biology* 37, 1838-1852. <https://doi.org/10.1016/j.ultrasmedbio.2011.08.004>.
- Atchley, A. A., and Prosperetti, A. (1989). The crevice model of bubble nucleation. *The Journal of the Acoustical Society of America* 86, 1065-1084. <https://doi.org/10.1121/1.398098>.
- Bhatnagar, S., Kwan, J. J., Shah, A. R., Coussios, C.-C., and Carlisle, R. C. (2016). Exploitation of sub-micron cavitation nuclei to enhance ultrasound-mediated transdermal transport and penetration of vaccines. *Journal of Controlled Release* 238, 22-30. <https://doi.org/10.1002/jps.23971>.
- Burgess, M., and Porter, T. (2015). On-demand cavitation from bursting droplets. *Acoustics Today* 11(4), 35-41.
- Carlisle, R., Choi, J., Bazan-Peregrino, M., Laga, R., Subr, V., Kostka, L., Ulbrich, K., Coussios, C.-C., and Seymour, L. W. (2013). Enhanced tumor uptake and penetration of virotherapy using polymer stealthing and focused ultrasound. *Journal of the National Cancer Institute* 105, 1701-1710. <https://doi.org/10.1093/jnci/djt305>.
- Coussios, C. C., Farny, C. H., ter Haar, G., and Roy, R. A. (2007). Role of acoustic cavitation in the delivery and monitoring of cancer treatment by high-intensity focused ultrasound (HIFU). *International Journal of Hyperthermia* 23, 105-120. <https://doi.org/10.1080/02656730701194131>.
- Coussios, C. C., and Roy, R. A. (2008). Applications of acoustics and cavitation to non-invasive therapy and drug delivery. *Annual Review of Fluid Mechanics* 40, 395-420. <https://doi.org/10.1146/annurev.fluid.40.111406.102116>.
- Couture, O., Hingot, V., Heiles, B., Muleki-Seya, P., and Tanter, M. (2018). Ultrasound localization microscopy and super-resolution: A state of the art. *IEEE Transactions on Ultrasonics Ferroelectrics and Frequency Control* 65, 1304-1320. <https://doi.org/10.1109/tuffc.2018.2850811>.
- Coviello, C., Kozick, R., Choi, J., Gyongy, M., Jensen, C., Smith, P., and Coussios, C. (2015). Passive acoustic mapping utilizing optimal beamforming in ultrasound therapy monitoring. *The Journal of the Acoustical Society of America* 137, 2573-2585. <https://doi.org/10.1121/1.4916694>.
- Dowling, A. P., and Ffowcs Williams, J. (1983). *Sound and Sources of Sound*. Ellis Horwood, Chichester, UK.
- Fox, F. E., and Herzfeld, K. F. (1954). Gas bubbles with organic skin as cavitation nuclei. *The Journal of the Acoustical Society of America* 26, 984-989. <https://doi.org/10.1121/1.1907466>.
- Gray, M. D., and Coussios, C. C. (2018). Broadband ultrasonic attenuation estimation and compensation with passive acoustic mapping. *IEEE Transactions on Ultrasonics Ferroelectrics and Frequency Control* 65, 1997-2011. <https://doi.org/10.1109/tuffc.2018.2866171>.
- Haworth, K. J., Bader, K. B., Rich, K. T., Holland, C. K., and Mast, T. D. (2017). Quantitative frequency-domain passive cavitation imaging. *IEEE Transactions on Ultrasonics Ferroelectrics and Frequency Control* 64, 177-191. <https://doi.org/10.1109/tuffc.2016.2620492>.
- Hockham, N. (2013). *Spatio-Temporal Control of Acoustic Cavitation During High-Intensity Focused Ultrasound Therapy*. PhD Thesis, University of Oxford, Oxford, UK.
- Hockham, N., Coussios, C. C., and Arora, M. (2010). A real-time controller for sustaining thermally relevant acoustic cavitation during ultrasound therapy. *IEEE Transactions on Ultrasonics Ferroelectrics and Frequency Control* 57, 2685-2694. <https://doi.org/10.1109/tuffc.2010.1742>.
- Holt, R. G., and Roy, R. A. (2001). Measurements of bubble-enhanced heating from focused, MHz-frequency ultrasound in a tissue-mimicking material. *Ultrasound in Medicine and Biology* 27, 1399-1412.
- Jensen, C., Cleveland, R., and Coussios, C. (2013). Real-time temperature estimation and monitoring of HIFU ablation through a combined modeling and passive acoustic mapping approach. *Physics in Medicine and Biology* 58, 5833. <https://doi.org/10.1088/0031-9155/58/17/5833>.
- Kennedy, J. E. (2005). High-intensity focused ultrasound in the treatment of solid tumours. *Nature Reviews Cancer* 5, 321-327. <https://doi.org/10.1038/nrc1591>.
- Khokhlova, V. A., Fowlkes, J. B., Roberts, W. W., Schade, G. R., Xu, Z., Khokhlova, T. D., Hall, T. L., Maxwell, A. D., Wang, Y. N., and Cain, C. A. (2015). Histotripsy methods in mechanical disintegration of tissue: towards clinical applications. *International Journal of Hyperthermia* 31, 145-162. <https://doi.org/10.3109/02656736.2015.1007538>.
- Konofagou, E. E. (2017). Trespassing the barrier of the brain with ultrasound. *Acoustics Today* 13(4), 21-26.
- Kwan, J. J., Graham, S., Myers, R., Carlisle, R., Stride, E., and Coussios, C. C. (2015). Ultrasound-induced inertial cavitation from gas-stabilizing nanoparticles. *Physical Review E* 92, 5. <https://doi.org/10.1103/PhysRevE.92.023019>.
- Leighton, T. G. (1994). *The Acoustic Bubble*. Academic Press, London, UK.
- Li, T., Khokhlova, T. D., Sapozhnikov, O. A., O'Donnell, M., and Hwang, J. H. (2014). A new active cavitation mapping technique for pulsed HIFU applications-bubble doppler. *IEEE Transactions on Ultrasonics Ferroelectrics and Frequency Control* 61, 1698-1708. <https://doi.org/10.1109/tuffc.2014.006502>.
- Lyon, P. C., Gray, M. D., Mannaris, C., Folkes, L. K., Stratford, M., Campo, L., Chung, D. Y. F., Scott, S., Anderson, M., Goldin, R., Carlisle, R., Wu, F., Middleton, M. R., Gleeson, F. V., and Coussios, C. C. (2018). Safety and feasibility of ultrasound-triggered targeted drug delivery of doxorubicin from thermosensitive liposomes in liver tumours (TARDOX): A single-centre, open-label, phase 1 trial. *The Lancet Oncology* 19, 1027-1039.
- Marmottant, P., and Hilgenfeldt, S. (2003). Controlled vesicle deformation and lysis by single oscillating bubbles. *Nature* 423, 153-156. <https://doi.org/10.1038/nature01613>.
- Matula, T. J., and Chen, H. (2013). Microbubbles as ultrasound contrast agents. *Acoustics Today* 9(1), 14-20.
- Molinari, M. (2012). *Mechanical Fractionation of the Intervertebral Disc*. PhD Thesis, University of Oxford, Oxford, UK.
- Myers, R., Coviello, C., Erbs, P., Follope, J., Rowe, C., Kwan, J., Crake, C., Finn, S., Jackson, E., Balloul, J.-M., Story, C., Coussios, C., and Carlisle, R. (2016). Polymeric cups for cavitation-mediated delivery of oncolytic vaccinia virus. *Molecular Therapy* 24, 1627-1633. <https://doi.org/10.1038/mt.2016.139>.
- Rapoport, N., Gao, Z. G., and Kennedy, A. (2007). Multifunctional nanoparticles for combining ultrasonic tumor imaging and targeted chemotherapy. *Journal of the National Cancer Institute* 99, 1095-1106. <https://doi.org/10.1093/jnci/djm043>.
- Rieke, V., and Pauly, K. (2008). MR thermometry. *Journal of Magnetic Resonance Imaging* 27, 376-390. <https://doi.org/10.1002/jmri.21265>.
- Stride, E. P., and Coussios, C. C. (2010). Cavitation and contrast: The use of bubbles in ultrasound imaging and therapy. *Proceedings of the Institution of Mechanical Engineers, Part H: Journal of Engineering in Medicine* 224, 171-191. <https://doi.org/10.1243/09544119jeim622>.
- Suslick, K. S. (1990). Sonochemistry. *Science* 247, 1439-1445. <https://doi.org/10.1126/science.247.4949.1439>.
- ter Haar, G., and Coussios, C. (2007). High intensity focused ultrasound: physical principles and devices. *International Journal of Hyperthermia* 23, 89-104. <https://doi.org/10.1080/02656730601186138>.



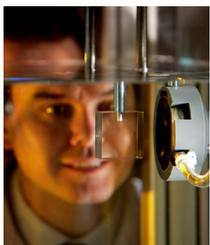
Michael D. Gray is a senior research fellow at the Biomedical Ultrasonics, Biotherapy and Biopharmaceuticals Laboratory (BUBBL), Institute of Biomedical Engineering, University of Oxford, Oxford, UK. His research interests include the use of sound, magnetism, and light

for targeted drug delivery; clinical translation of cavitation monitoring techniques; and hearing in marine animals. He holds a PhD from the Georgia Institute of Technology (2015) and has over 26 years of experience in acoustics applications ranging from cells to submarines.



Eleanor Stride is a professor of biomaterials at the University of Oxford, Oxford, UK, specializing in stimuli responsive drug delivery. She obtained her BEng and PhD in mechanical engineering from University College London, UK, before moving to Oxford in 2011. She

has published over 150 academic papers, has 7 patents, and is a director of 2 spinout companies set up to translate her research into clinical practice. Her contributions have been recognized with several awards, including the 2015 Acoustical Society of America (ASA) Bruce Lindsay Award and the Institution of Engineering and Technology (IET) A. F. Harvey Prize. She became a fellow of the Royal Academy of Engineering in 2017 and of the ASA in 2018.



Constantin-C. Coussios is the director of the Institute of Biomedical Engineering and the holder of the first statutory chair in Biomedical Engineering at the University of Oxford, Oxford, UK, where he founded and heads the Biomedical Ultrasonics, Biotherapy and Biopharmaceuticals Laboratory (BUBBL) since 2004. He is a recipient of the Acoustical Society of America (ASA) F. V. Hunt Postdoctoral Fellowship (2002), the ASA Bruce Lindsay Award in 2012, and the Silver Medal from the UK Royal Academy of Engineering in 2017 and has been a fellow of the ASA since 2009. He holds BA, MEng, and PhD (2002) degrees from the University of Cambridge, Cambridge, UK and has cofounded two medical device companies that exploit acoustic cavitation for minimally invasive surgery (OrthoSon Ltd.) and oncological drug delivery (OxSonics Ltd.).

NGC[®] TESTING SERVICES

ACOUSTICAL • FIRE • STRUCTURAL • ANALYTICAL™

Over 50 Years of Test Services

NVLAP Accredited (Code 200291-0)

IAS Accredited (Lab Code 216)

ISO/IEC 17025 Compliant

Test Capabilities Include:

E492 IIC, E2179 Delta IIC E90 STC

C423 NRC, E1414 CAC, E1111 AC

Sound Power, Pipe Lagging, SAE J1400

E84 Flame Spread, E119 Fire Endurance

NGC Testing Services • 1650 Military Road

Buffalo, NY 14217 • Tel: 716-873-9750

ngctestingservices.com

ASA SCHOOL 2020

Living in the Acoustic Environment

May 9-10, 2020 • Itasca, IL

- **Two-Day Program** Lectures, demonstrations, and discussions by distinguished acousticians covering interdisciplinary topics in eight technical areas
- **Participants** Graduate students and early career acousticians in all areas of acoustics
- **Location** Eaglewood Resort and Spa, Itasca, IL
- **Dates** May 9-10, 2020, immediately preceding the ASA spring meeting in Chicago
- **Cost** \$50 registration fee, which includes hotel, meals, and transportation from Eaglewood to the ASA meeting location
- **More Information** Application form, preliminary program, and more details will be available in November, 2019 at AcousticalSociety.org



The Art of Concert Hall Acoustics: Current Trends and Questions in Research and Design

Kelsey A. Hochgraf

Address:
Acentech
33 Moulton Street
Cambridge, Massachusetts
02138
USA

Email:
khochgraf@acentech.com

Concert hall design exists at the intersection of art, science and engineering, where acousticians continue to demystify aural excellence.

What defines “excellence” in concert hall acoustics? Acousticians have been seeking perceptual and physical answers to this question for over a century. Despite the wealth of insightful research and experience gained in this time, it remains established canon that the best concert halls for classical orchestral performance are the Vienna Musikverein (1870), Royal Concertgebouw in Amsterdam (1888), and Boston Symphony Hall (1900; Beranek, 2004). Built within a few decades of each other, the acoustical triumph of these halls is largely attributable to their fortuitous “shoebox” shape and emulation of other successful halls. Today, we have a significantly more robust understanding of how concert halls convey the sounds of musical instruments, and we collect tremendous amounts of perceptual and physical data to attempt to explain this phenomenon, but in many respects, the definition of excellence remains elusive.

This article discusses current trends in concert hall acoustical design, including topics that are well understood and questions that have yet to be answered, and challenges the notion that “excellence” can be defined by a single room shape or set of numerical parameters.

How Should a Concert Hall Sound?

This is the fundamental question asked at the outset of every concert hall project, but it is surprisingly difficult to answer succinctly. The primary purpose of a concert hall is to provide a medium for communication between musicians and the audience (Blauert, 2018). There are several percepts of the acoustical experience, different for musicians and listeners, that are critical for enabling this exchange. On stage, musicians need good working conditions so that they hear an appropriate balance of themselves, each other, and the room. For a listener in the audience, articulating the goals is more difficult.

Listeners want to be engaged actively by the music, but the acoustical implications of this goal are complex and highly subjective. This question has been the focus of rich and diverse research for decades, including notable contributions by Beranek (1962, 1996, 2004; summarized in a previous issue of *Acoustics Today* by Markham, 2014), Hawkes and Douglas (1971), Schroeder et al. (1974), Soulodre and Bradley (1995), and Lokki et al. (2012) among others. These studies have established a common vocabulary of relevant perceptual characteristics and have attempted to distill the correlation between listener preference and perception to a few key factors, but it remains true that acoustical perception in concert halls is multidimensional. Kuusinen and Lokki (2017) recently proposed a “wheel of concert hall acoustics,”

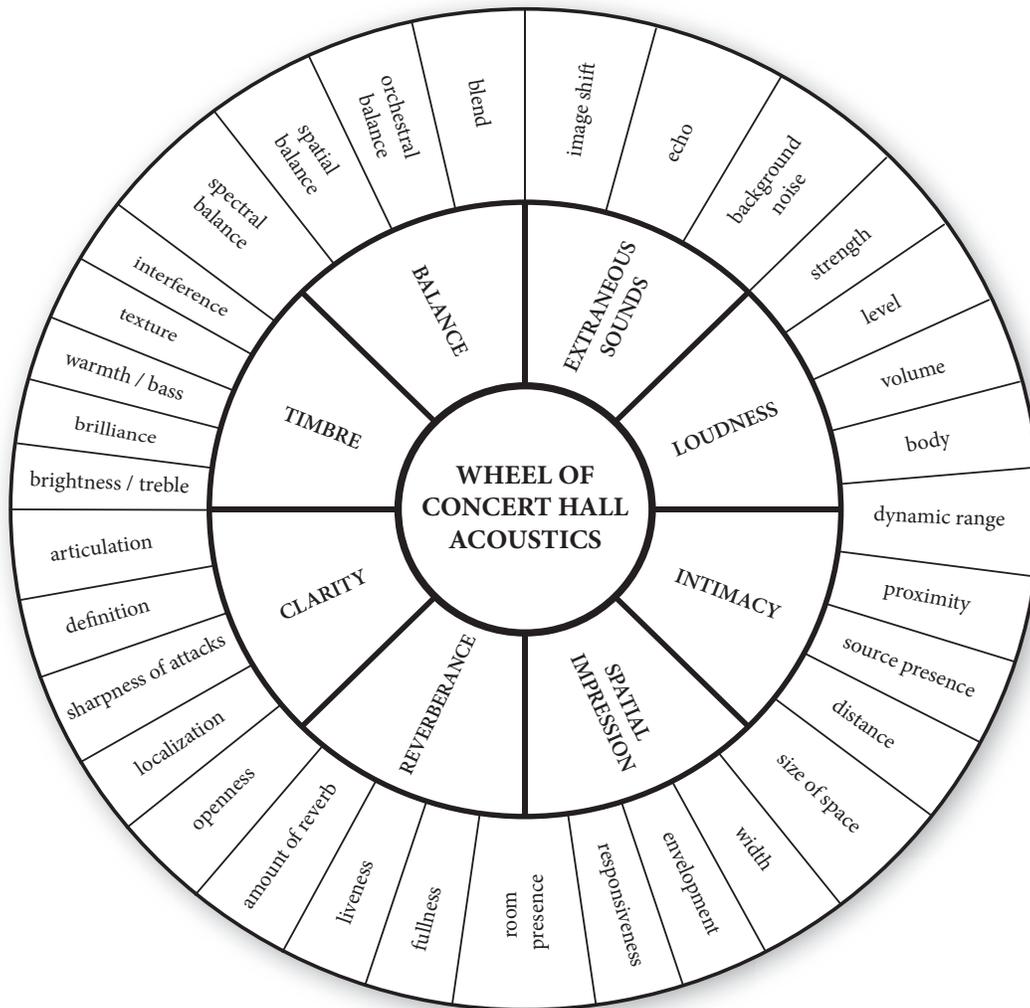


Figure 1. “Wheel of concert hall acoustics.” The graphic proposes a common vocabulary for eight primary attributes (inner ring) of acoustical perception in concert halls, and several related sub-factors (outer ring). Some attributes in the outer ring overlap between primary percepts, illustrating their interdependency. The circular organization highlights the fact that there is not a consensus hierarchy of characteristics correlated with listener preference, and the structure does not assume orthogonality between any pair of perceptual attributes. From Kuusinen and Lokki (2017), with permission from S. Hirzel Verlag.

shown in **Figure 1**, that groups relevant perceptual factors into eight categories: clarity, reverberance, spatial impression, intimacy, loudness, balance, timbre, and minimization of extraneous sounds.

Although there is not yet a consensus around specific attributes most correlated with audience listener preference, there is agreement that different people prioritize different elements of the acoustical experience. Several studies have shown that listeners can be categorized into at least two preference groups: one that prefers louder, more reverberant and enveloping acoustics and another that prefers a more intimate and clearer sound (Lokki et al., 2012; Beranek, 2016). The listening preferences of the first category generally align

with the acoustical features of “shoebox” concert halls (tall, narrow, and rectangular), such as those in Vienna, Boston, and Amsterdam. Among other factors, the second category of listeners may be influenced by perceptual expectations developed from listening to recordings of classical orchestral music rather than attending live performances (Beranek et al., 2011). However, all elements remain important to the listening experience. Even listeners who prefer a clearer sound still require an adequate level of loudness, reverberance, and envelopment (Lokki et al., 2012). Historically, acousticians have considered some percepts, such as reverberance and clarity, to be in direct opposition with each other, but there is new emphasis on finding a common ground to engage more listeners.

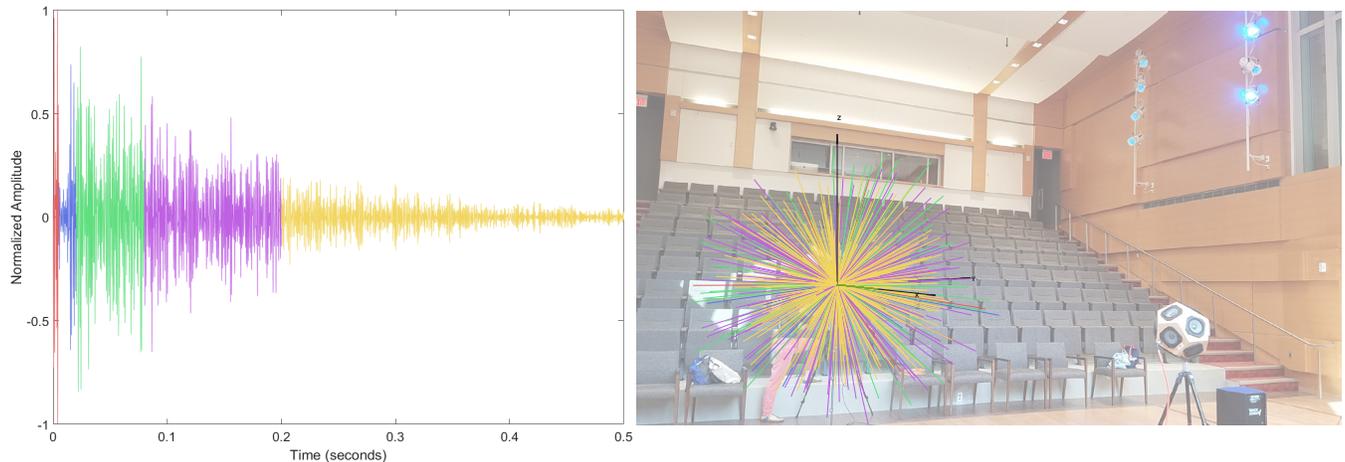


Figure 2. Measured impulse responses (IRs) from the Rachel Carson Music and Campus Center Recital Hall, Concord, MA. Both IRs were measured with an omnidirectional (dodecahedral) loudspeaker, band-pass filtered in a 1-kHz octave band, and color coded by temporal region. **Left:** IR, measured with an omnidirectional microphone, shows relative timing and level of reflections arriving from all directions. **Right:** IR, measured with a multichannel directional microphone, shows relative level and spatial distribution of reflections. Photo by Kelsey Hochgraf.

Using Auditory Stream Segregation to Decode a Musical Performance

One innovative approach is based on the principles of auditory stream segregation, building on Bregman’s (1990) model of auditory scene analysis. According to this model, the brain decomposes complex auditory scenes into separate streams to distinguish sound sources from each other and from the background noise. The “cocktail party effect” is a common example of auditory stream segregation, which describes how a person can selectively focus on a single conversation in a noisy room yet still subconsciously process auditory information from the noise (e.g., the listener will notice if someone across the room says their name). See the discussion of similar issues in classrooms in Leibold’s article in this issue of *Acoustics Today*.

In a concert hall, the listener’s brain is presented with a complex musical scene that needs to be organized in some way to extract meaning; otherwise, it would be perceived as noise. Supported by research studies at the Institute for Research and Coordination in Acoustics and Music in Paris, Kahle (2013) and others have suggested that the brain decomposes the auditory scene in a concert hall into distinct streams: the source and the room. If listeners can perceive the source separately from the room, then they can perceive clarity of the orchestra while simultaneously experiencing reverberance from the room. Griesinger (1997) has suggested that this stream segregation is contingent on the listener’s ability to localize the direct sound from individual instruments separately from other

instruments and from reflections in the room. Although the brain may perceive the auditory streams separately, the source and room responses are dependent on each other acoustically. Developing a better understanding of this relationship, both spatially and temporally, is critical to integrating the range of acoustical percepts more holistically in the future.

Setting Acoustical Goals for a New Concert Hall

Without one set of perceptual factors to guarantee acoustical excellence, who determines how a new concert hall should sound? In recent interviews between the author of this article and acousticians around the world, three typical answers emerged: the orchestra, the acoustician, or a combination of both. Scott Pfeiffer, Robert Wolff, and Alban Bassuet discussed the early design process for new concert halls in the United States, which often includes visiting existing spaces so that the orchestra musicians, conductor, acoustician, and architect can listen to concerts together and discuss what they hear. Pfeiffer (personal communication, 2018) expressed the value of creating a “shared language with clients to allow them to steer the acoustic aesthetic.” Wolff (personal communication, 2018) mentioned that orchestra musicians often have strong acoustical preferences developed from playing with each other for many years and having frequent opportunities to listen to each other from an audience perspective. Bassuet (personal communication, 2018) asserted that it is more the acoustician’s responsibility to “transpose into the musician’s head when designing the hall” and set the perceptual goals accordingly.

Table 1. Summary of standard room acoustics parameters

Parameter	Description	Intended Perceptual Correlate
Reverberation time (RT, T ₂₀ , T ₃₀)	Time for sound to decrease by 60 dB, based on linear fit to energy decay (excluding direct sound and earliest reflections)	N/A
Early decay time (EDT)	Similar to RT, but based on initial 10 dB of decay (including direct sound and earliest reflections)	Reverberance
Strength (G)	Logarithmic ratio between sound energy in room vs. free field 10-m away from the same source	Loudness
Clarity (C ₈₀)	Logarithmic ratio between early (0-80 ms) and late (after 80 ms) energy	Clarity
Early lateral energy fraction (J _{LF})	Ratio between lateral and total energy within first 80 ms	Apparent source width
Late lateral sound level (L _l)	Logarithmic ratio between late lateral energy (after 80 ms) and total energy	Listener envelopment
Interaural cross-correlation coefficient (IACC _{Early} , IACC _{Late})	Binaural measure of similarity between sound at left and right ears, reported separately for early (0-80 ms) and late (after 80 ms) energy	Early: apparent source width Late: listener envelopment
Stage support (ST _{Early} , ST _{Late})	Ratios between reflected and direct sound measured on stage, reported separately for early (20-100 ms) and late (100-1,000 ms) energy	Early: ensemble hearing Late: reverberance on stage

Data from the International Organization for Standardization (ISO, 2009).

Eckhard Kahle (personal communication, 2018) discussed how the early design process differs outside the United States, where project owners often hire independent acousticians to develop an “acoustic brief,” and design teams compete against each other to develop a conceptual design that most effectively responds to the acoustical goals outlined by the brief. With such a variety of perceptual factors and approaches to prioritizing them, it is no surprise that concert halls around the world sound as different from each other as they do.

How Can Perceptual Goals be Translated into Quantitative, Measurable Data?

Considering a concert hall as a linear, time-invariant system, an impulse-response measurement can be used to understand how the room modifies the sounds of musical instruments. **Figure 2** shows impulse responses measured with an omnidirectional loudspeaker and two different microphones. The impulse response measured with an omnidirectional microphone (**Figure 2, left**), illustrates the direct sound path from the loudspeaker to the microphone, early reflections that are strong and distinct from each other and weaker late reflections that occur closely spaced in time and decay smoothly. Measurements are analyzed by octave band

and divided temporally, although the time periods most relevant to each percept are the subject of ongoing research. To understand the spatial distribution of reflections around the listener, a multi-channel impulse response can be measured with a directional microphone array (**Figure 2, right**). Several numerical parameters standardized by ISO 3382-1:2009 (International Organization for Standardization [ISO], 2009; summarized in **Table 1**), can be derived from impulse responses measured with an omnidirectional sound source.

There is growing consensus among acousticians that although many of these parameters are useful, they do not provide a complete representation of concert hall acoustics. In an interview with the author of

this article, Pfeiffer (personal communication, 2018) noted “there’s a range of performance in any of the given parameters that’s acceptable, and a range that’s not” but that the “mythical holy grail of perfect acoustics” does not exist. Acoustician Paul Scarbrough (personal communication, 2018) noted that standard parameters are particularly limited in describing the spatial distribution of sound, saying “we’re not measuring the right things yet.” Bassuet (personal communication, 2018) suggested that “we should not be afraid to make connections between emotional value and [new] acoustical metrics.” Recent article titles are similarly critical and illustrative: “In Search of a New Paradigm: How Do our Parameters and Measurement Techniques Constrain Approaches to Concert Hall Design” (Kirkegaard and Gulsrund, 2011) and “Throw Away that Standard and Listen: Your Two Ears Work Better” (Lokki, 2013).

Deficiencies of Existing Objective Parameters

In summary, the limitations are largely attributable to differences between an omnidirectional sound source and an orchestra and between omnidirectional microphones and the human hearing system. As described in detail by Meyer (2009), each musical instrument has unique and frequency-dependent radiation characteristics. As musicians vary their

dynamics, the spectrum also changes; at higher levels, more overtones are present. These nonlinear radiation characteristics are not represented by an omnidirectional sound source, and the directivity of the sound source has a significant impact on perception of room acoustics (Hochgraf et al., 2017). This problem becomes more pronounced, complicated, and perceptually important when considering the radiation of an entire orchestra.

In the audience, the listener hears the orchestra spatially, not monaurally. Binaural and directional parameters for spatial impression are helpful, but measurement of these parameters can be unreliable due to the impacts of microphone calibration and orientation on the parametric calculation. In addition to directivity mismatches between real sources and receivers and those used to measure impulse responses, the frequency ranges of human hearing and radiation from musical instruments are both significantly larger than the capabilities of most loudspeakers used for room acoustics measurements. Beyond the inherent limitations of the standard parameters, impulse responses are often measured in unoccupied rooms and averaged across multiple seats. Occupied rooms, however, are significantly different acoustically from unoccupied rooms, and acoustical conditions vary significantly between different seats.

As the significance of these limitations has become clearer, acousticians have developed new ways of analyzing impulse responses to obtain useful information. More emphasis is being placed on visual and aural inspection of the impulse response instead of on parameters that can obscure important acoustical details. Acousticians also modify standard parameters in a variety of ways, including changing temporal increments, separating calculation for early and late parts of the impulse response, and averaging over different frequency ranges. Although these adaptations may help an individual acoustician make sense of data, it makes it difficult to compare halls with each other—one of the primary reasons for documenting parameters in the first place.

New parameters are also emerging, including Griesinger's (2011) localization metric (LOC) for predicting the ability of a listener to detect the position of a sound source and Bassuet's (2011) ratios between low/high (LH) and front/rear (FR) lateral energy arriving at the sides of a listener's head. The potential utility of objective metrics that correlate well with perception is undeniable, but as more parameters emerge and their application among acousticians diverges, we are continually faced by the question of whether parameters fundamentally support or suppress excellent design.



Figure 3. Boston Symphony Hall. Photo by Peter Vanderwarker.



Figure 4. Philharmonie de Paris. Photo by Jonah Sacks.

How Do Perceptual Goals and Parametric Criteria Inform Design?

The form of a concert hall is determined by a variety of factors, including, but not limited to, acoustics. One of the greatest challenges and responsibilities of an acoustician is to educate the design team of the implications of room shape and size, which fundamentally determine the range of possible acoustical outcomes, so that design decisions are well-informed and consistent with the perceptual goals.

Case Studies of Two Concert Halls:

The Influence of Shape, Size, and Parametric Criteria

Boston Symphony Hall (Figure 3) and the Philharmonie de Paris (Figure 4) were built over a century apart, and although they are used for the same purpose, they are fundamentally different acoustically.

Constructed in 1900, the shoebox shape of Boston Symphony Hall is the result of architectural evolution from royal courts, ballrooms, and the successful, similarly shaped Gewandhaus Hall in Leipzig, Germany (Beranek, 2004). It was the first hall built with any quantitative acoustical design input, courtesy of Wallace Clement Sabine and his timely dis-

covery of the relationship between room volume, materials, and reverberation. Built in 2015, the Philharmonie de Paris is far from rectangular. Its form is most similar to a “vineyard” style hall, which features a centralized orchestra position surrounded by audience seated in shallow balconies. Led by Jean Nouvel (architect) and Harold Marshall (design acoustician), the design team was selected by competition and was provided with a detailed and prescriptive acoustical brief by the owner’s acoustician (Kahle Acoustics and Altia, 2006). Sabine’s scientifically based acoustical design input for Boston Symphony Hall was limited to its ceiling height. On the other hand, the design of the Philharmonie de Paris was based on decades of research about concert hall acoustics. How do these halls, built under such different circumstances, compare with each other acoustically?

Boston Symphony Hall is known for its generous reverberance, warmth, and enveloping sound. Its 2,625 seats are distributed across a shallowly raked floor and two balconies that wrap around the sides of the room, which measures 18,750 m³ in total volume. If the seats were rebuilt today, code requirements and current expectations of comfort would significantly reduce the seat count. The lightly upholstered seats are acoustically important for preserving strength and reverberance (Beranek, 2016). Heavy, plaster walls reflect low-frequency energy and help to create a warm timbre. The hard and flat lower side walls and undersides of shallow side balconies provide strong lateral reflections to the orchestra seating level, and statues along the ornamented upper side walls and deeply coffered ceiling promote diffuse late reflections throughout the room. The large volume above the second balcony fosters the development of late reverberation that is long, relatively loud, and decays smoothly. The perception of clarity in the room varies by listening position and the music being performed. The Boston Symphony Orchestra is also one of the world’s best orchestras, and it knows how to highlight the hall’s acoustical features, particularly for late classical and romantic era repertoire.

The Philharmonie de Paris seats 2,400 in a total volume of 30,500 m³, which is over 60% larger than Boston Symphony Hall. One of the results of this significant difference in size is that although the Philharmonie’s reverberation

is long in time, it is lower in level, which strikes a different balance between reverberance and clarity. In an interview with Kahle (personal communication, 2018), he noted that the high seat count was one of the primary acoustical design challenges and this necessitated the parametrically prescriptive design brief. In his words, “an orchestra has a limited sound power, which you have to distribute over more people...and if you share it, you get less of it.” The seating arrangement keeps the audience closer to the musicians, which heightens the sense of intimacy. Lateral reflections are provided by balcony fronts, although the distribution of lateral energy and ensemble balance vary more between seats due to the shape of the room and position of the orchestra. Concave wall surfaces are shaped to scatter sound and avoid focusing. The lengthy but relatively less loud reverberation is generated by an outer volume that is not visible to the audience. The same orchestra playing the same repertoire will sound completely different in Paris compared with Boston, and the quality of the listening experience will depend on where one is sitting, the music being performed, and, most of all, the listener’s expectations and preferences.

Table 2 shows a comparison of parameters measured in both halls. The numbers show some interesting relative differences but do not convey the perceptual significance of these differences or predict how someone would perceive the acoustics of either hall from a particular seat. Standard deviation from the average measured parameters should be at least as important as the averages, especially for spatial parameters, although this information is rarely published. None of the measured parameters describe timbre, ensemble balance, or blend. Although the parameters do impart meaning, especially in the context of listening observations and an understanding of how architectural features in the room impact the acoustics, they do not describe the complete acoustical story or provide a meaningful account of the halls’ dramatic differences.

Table 2. Average mid-frequency parameters measured in Boston Symphony Hall and Philharmonie de Paris

	RT _{unocc} , s	RT _{occ} , s	EDT, s	G _{mid} , dB	C ₈₀ , dB	J _{LF}
Boston Symphony Hall	2.5	1.9	2.4	4.0	-2.6	0.24
Philharmonie de Paris	3.2	2.5	Not reported	2.2	-0.7	0.20

Data for Boston Symphony Hall from Beranek (2004) and for Philharmonie de Paris from Scelo et al. (2015).

Where do we go from here? It can be tempting to conclude that all new concert halls should be shoebox shaped and that the acoustics in more complex geometries remain an unsolvable mystery. But as long as architectural and public demand for creative room shapes continues to grow, we must keep pursuing answers.

Two Emerging Design Trends from Recent Research and Experience

The importance of lateral reflections for spatial impression is well-understood (Barron, 1971; Barron and Marshall, 1981; Lokki et al., 2011), but more recent research has shown that these reflections are also critical to the perception of dynamic responsiveness. Pätynen et al. (2014) have recently shown that lateral reflections increase the perceived dynamic range by emphasizing high-frequency sounds as the result of two important factors: musical instruments radiate more high-frequency harmonics as they are played louder and the human binaural hearing system is directional and more sensitive to high-frequency sound that arrives from the sides. If lateral reflections are present, they will emphasize high-frequency sound radiated from instruments as they crescendo, and our ears, in turn, will emphasize these same frequencies. If they are not present, then the perceived dynamic range will be more limited. Increased perception of dynamic range has also been shown to correlate with increased emotional response (Pätynen and Lokki, 2016).

Building on these developments, Green and Kahle (2018) have recently shown that the perception threshold for lateral reflections decreases with increasing sound level, meaning that more lateral reflections will be perceived by the listener as the music crescendos, further heightening the sense of dynamic responsiveness. From an acoustical design perspective, it is easier to provide strong lateral reflections for a larger audience area in a shoebox hall by simply leaving large areas of the lower side wall surfaces hard, massive, and flat. In a vineyard hall, the design process is more difficult because individual balcony fronts and side wall surfaces are smaller and less evenly impact the audience area.

The role of diffusion has been hotly debated in architectural acoustics for a long time. A diffuse reflection is weaker than a specular reflection and scatters sound in all directions. Diffusion is helpful for avoiding problematic reflection patterns (such as echoes or focusing effects) without adding unwanted sound absorption. It can also be helpful for creating a more uniform late sound field (such as in the upper volume of Boston Symphony Hall). Haan and Fricke (1997) studied

the correlation between estimated surface diffusivity and overall acoustical quality perceived by musicians playing in 53 different halls. As a result of the high correlation that they found, as well as design preferences of many acousticians and architects at the time, many halls built in the last two decades have a high degree of surface diffusivity. Not all of these halls have been regarded as acoustically successful, particularly when the articulation has all been at the same physical scale (meaning that surfaces diffuse sound in a narrow range of frequencies) and when the diffusion has weakened lateral reflections that we now better understand to be critical to multiple perceptual factors.

The title of a recent presentation is particularly illustrative of the growing opinion among acousticians who caution against the use of too much diffusion: “Halls without qualities – or the effect of acoustic diffusion” (Kahle, 2018). Although the tide seems to be shifting away from high surface diffusivity and there is more evidence to substantiate the need for strong lateral reflections, there is still limited evidence from research to explain exactly how diffusion impacts the listening experience.

How Will Concert Hall Acoustical Design Change in the Future?

In parallel with applying lessons learned from existing halls, the future of concert hall acoustical design will be transformed by the power of auralization. An auralization is an aural rendering of a simulated or measured space, created by convolving impulse responses with anechoic audio recordings, played over loudspeakers or headphones for spatially realistic listening. Auralizations have been used in research and limited design capacities for several years, but recent technological advancements associated with measurement, simulation, and spatial audio have the potential to leverage auralization for more meaningful and widespread use in the future, potentially answering previously unresolved questions about concert hall acoustical perception and design. Rather than averaging and reducing impulse responses to single number parameters, auralizations strive to preserve all the perceptually important complexities and allow acousticians to make side-by-side comparisons with their best tools: their ears.

Auralizing the design of an unbuilt space requires simulating its impulse response. Commercially available room acoustics software currently relies on geometric simulation methods that model sound waves as rays, which is a valid ap-

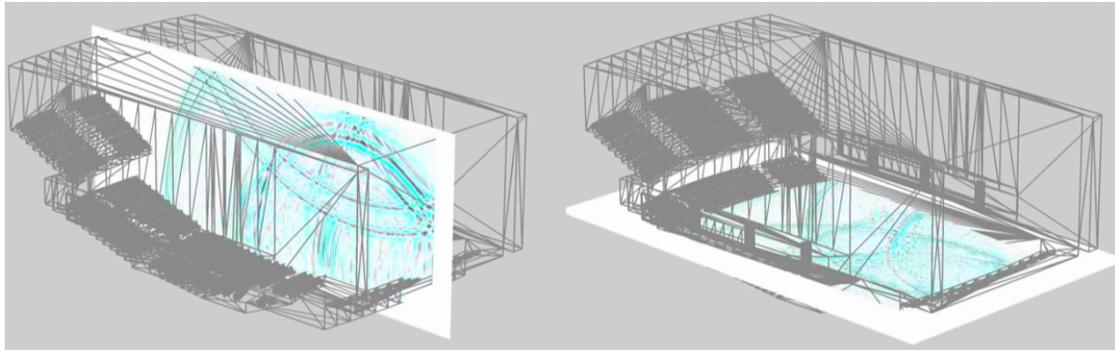


Figure 5. Screenshot from wave-based simulation of sound propagation in the Experimental Media and Performing Arts Center Concert Hall, Troy, NY.

proximation for asymptotically high frequencies, when the wavelength of sound is much smaller than the dimensions of room surfaces but not for low frequencies or wave behaviors such as diffusion and diffraction. Wave-based modeling requires approximating the solution to the wave equation, typically using finite volume, finite element, or boundary element methods. These methods have existed for many years, but computational complexity has limited widespread use in concert hall acoustical design. **Figure 5** shows a screenshot from a wave-based simulation, modeled as part of a research effort to highlight its potential utility in concert hall acoustics (Hochgraf, 2015). By harnessing the computing power of parallelized finite-volume simulations over multiple cloud-based graphics-processing units (GPUs), wave-based modeling may become widely available and computationally efficient very soon, allowing acousticians to test their designs with more accuracy and reliability (Hamilton and Bilbao, 2018).

An auralization will never replace the real experience of listening to music in a concert hall because it does not enable direct, engaging communication between musicians and listeners. As a musician and frequent audience member myself, I look forward to more opportunities in the future to draw from these real listening experiences and to use auralization as a research and design tool to support innovative, “excellent” design.

Acknowledgments

I thank Alban Bassuet, Timothy Foulkes, Eckhard Kahle, Scott Pfeiffer, Rein Pirn, Paul Scarbrough, and Robert Wolff for sharing their candid thoughts in interviews on concert hall acoustical design. I am also especially grateful to Jonah Sacks and Ben Markham for their feedback and mentorship.

References

- Barron, M. (1971). The subjective effects of first reflections in concert halls: The need for lateral reflections. *Journal of Sound and Vibration* 15(4), 475-494.
- Barron, M., and Marshall, H. (1981). Spatial impression due to early lateral reflections in concert halls: The derivation of a physical measure. *Journal of Sound and Vibration* 77(2), 211-232.
- Bassuet, A. (2011). New acoustical parameters and visualization techniques to analyze the spatial distribution of sound in music spaces. *Building Acoustics* 18(3-4), 329-347. <https://doi.org/10.1260/1351-010X.18.3-4.329>.
- Beranek, L. L. (1962). *Music, Acoustics, and Architecture*. John Wiley & Sons, New York.
- Beranek, L. L. (1996). *Concert Halls and Opera Houses: How They Sound*. American Institute of Physics for the Acoustical Society of America, Woodbury, NY.
- Beranek, L. L. (2004). *Concert Halls and Opera Houses: Music, Acoustics, and Architecture*, 2nd ed. Springer-Verlag, New York.
- Beranek, L. L. (2016). Concert hall acoustics: Recent findings. *The Journal of the Acoustical Society of America* 139, 1548-1556. <https://doi.org/10.1121/1.4944787>.
- Beranek, L. L., Gade, A. C., Bassuet, A., Kirkegaard, L., Marshall, H., and Toyota, Y. (2011). Concert hall design—present practices. *Building Acoustics* 18(3-4), 159-180. (This is a condensation of presentations at a special session on concert hall acoustics at the International Symposium on Room Acoustics, Melbourne, Australia, August 29-31, 2010.)
- Blauert, J. (2018). Assessing “quality of the acoustics” at large. *Proceedings of the Institute of Acoustics: Auditorium Acoustics*, Hamburg, Germany, October 4-6, 2018.
- Bregman, A. S. (1990). *Auditory Scene Analysis*. The MIT Press, Cambridge, MA.
- Green, E., and Kahle, E. (2018). Dynamic spatial responsiveness in concert halls. *Proceedings of the Institute of Acoustics: Auditorium Acoustics*, Hamburg, Germany, October 4-6, 2018.
- Griesinger, D. (1997). The psychoacoustics of apparent source width, spaciousness and envelopment in performance spaces. *Acta Acustica united with Acustica* 83(4), 721-731.
- Griesinger, D. (2011). The relationship between audience engagement and the ability to perceive pitch, timbre, azimuth and envelopment of multiple sources. *Proceedings of the Institute of Acoustics: Auditorium Acoustics*, Dublin, Ireland, May 20-22, 2011, vol. 33, pp. 53-62.
- Haan, C., and Fricke, F. (1997). An evaluation of the importance of surface diffusivity in concert halls. *Applied Acoustics* 51(1), 53-69.

- Hamilton, B., and Bilbao, S. (2018). Wave-based room acoustics modelling: Recent progress and future outlooks. *Proceedings of the Institute of Acoustics: Auditorium Acoustics*, Hamburg, Germany, October 4-6, 2018.
- Hawkes, R. J., and Douglas, H. (1971). Subjective acoustic experience in concert auditoria. *Acta Acustica united with Acustica* 24(5), 235-250.
- Hochgraf, K. (2015). *Auralization of Concert Hall Acoustics Using Finite Difference Time Domain Methods and Wave Field Synthesis*. MS Thesis, Rensselaer Polytechnic Institute, Troy, NY.
- Hochgraf, K., Sacks, J., and Markham, B. (2017). The model versus the room: Parametric and aural comparisons of modeled and measured impulse responses. *The Journal of the Acoustical Society of America* 141(5), 3857. <https://doi.org/10.1121/1.4988612>.
- International Organization for Standardization (ISO). (2009). *Acoustics – Measurement of Room Acoustics Parameters – Part 1: Performance Spaces*. International Organization for Standardization, Geneva, Switzerland.
- Kahle Acoustics and Altia. (2006). *Philharmonie de Paris Acoustic Brief, Section on Concert Hall Only*. Available at http://www.kahle.be/articles/AcousticBrief_PdP_2006.pdf. Accessed October 30, 2018.
- Kahle, E. (2013). Room acoustical quality of concert halls: Perceptual factors and acoustic criteria—Return from experience. *Building Acoustics* 20(4), 265-282.
- Kahle, E. (2018). Halls without qualities - or the effect of acoustic diffusion. *Proceedings of the Institute of Acoustics: Auditorium Acoustics*, Hamburg, Germany, October 4-6, 2018.
- Kirkegaard, L., and Gulsrund, T. (2011). In search of a new paradigm: How do our parameters and measurement techniques constrain approaches to concert hall design? *Acoustics Today* 7(1), 7-14.
- Kuusinen, A., and Lokki, T. (2017). Wheel of concert hall acoustics. *Acta Acustica united with Acustica* 103(2), 185-188. <https://doi.org/10.3813/AAA.919046>.
- Lokki, T. (2013). Throw away that standard and listen: Your two ears work better. *Journal of Building Acoustics* 20(4), 283-294. <https://doi.org/10.1260/1351-010X.20.4.283>.
- Lokki, T., Pätynen, J., Kuusinen, A., and Tervo, S. (2012). Disentangling preference ratings of concert hall acoustics using subjective sensory profiles. *The Journal of the Acoustical Society of America* 132, 3148-3161. <https://doi.org/10.1121/1.4756826>.
- Lokki, T., Pätynen, J., Tervo, S., Siltanen, S., and Savioja, L. (2011). Engaging concert hall acoustics is made up of temporal envelope preserving reflections. *The Journal of the Acoustical Society of America* 129(6), EL223-EL228. <https://doi.org/10.1121/1.3579145>.
- Markham, B. (2014). Leo Beranek and concert hall acoustics. *Acoustics Today* 10(4), 48-58.
- Meyer, J. (2009). *Acoustics and the Performance of Music: Manual for Acousticians, Audio Engineers, Musicians, Architects and Musical Instrument Makers*, 5th ed. Springer-Verlag, New York. Translated by U. Hansen.
- Pätynen, J., and Lokki, T. (2016). Concert halls with strong and lateral sound increase the emotional impact of orchestra music. *The Journal of the Acoustical Society of America* 139(3), 1214-1224. <https://doi.org/10.1121/1.4944038>.
- Pätynen, J., Tervo, S., Robinson, P., and Lokki, T. (2014). Concert halls with strong lateral reflections enhance musical dynamics. *Proceedings of the National Academy of Sciences of the United States of America* 111(12), 4409-4414. <https://doi.org/10.1073/pnas.1319976111>.
- Scelo, T., Exton, P., and Day, C. (2015). Commissioning of the Philharmonie de Paris, Grande Salle. *Proceedings of the Institute of Acoustics: Auditorium Acoustics*, Paris, October 29-31, 2015, vol. 37, pp. 128-135.
- Schroeder, M., Gottlob, G., and Siebrasse, K. (1974). Comparative study of European concert halls: Correlation of subjective preference with geometric and acoustics parameters. *The Journal of the Acoustical Society of America* 56, 1195-1201. <https://doi.org/10.1121/1.1903408>.
- Soulodre, G. A., and Bradley, J. (1995). Subjective evaluation of new room acoustic measures. *The Journal of the Acoustical Society of America* 98, 294-301. <https://doi.org/10.1121/1.413735>.

BioSketch



Kelsey Hochgraf is a senior consultant in the architectural acoustics group at Acentech, Cambridge, MA. She works on a variety of interdisciplinary projects but has a particular interest in the performing arts and educational facilities as well as in acoustical modeling and auralization. Kelsey also teaches an acoustics class in the mechanical engineering department at Tufts University, Medford, MA. She holds a BSE in mechanical and aerospace engineering from Princeton University, Princeton, NJ, and an MS in architectural acoustics from Rensselaer Polytechnic Institute, Troy, NY. Find out more about Kelsey's interests and background in "Ask an Acoustician" in the winter 2017 issue of *Acoustics Today*.

Women in Acoustics

The ASA's Women in Acoustics Committee was created in 1995 to address the need to foster a supportive atmosphere within the Society and within the scientific community at large, ultimately encouraging women to pursue rewarding and satisfying careers in acoustics.

Learn more about the committee at
<http://womeninacoustics.org>.

Too Young for the Cocktail Party?

One reason why children and cocktail parties do not mix.

Lori J. Leibold

Address:

Center for Hearing Research
Boys Town National Research Hospital
555 North 30th Street
Omaha, Nebraska 68131
USA

Email:

lori.leibold@boystown.org

Emily Buss

Address:

Department of Otolaryngology/Head
and Neck Surgery
University of North Carolina at Chapel
Hill
170 Manning Drive
Campus Box 7070
Chapel Hill, North Carolina 27599
USA

Email:

ebuss@med.unc.edu

Lauren Calandruccio

Address:

Department of Psychological Sciences
Case Western Reserve University
Cleveland Hearing and Speech Center
11635 Euclid Avenue
Cleveland, Ohio 44106
USA

Email:

lauren.calandruccio@case.edu

There are many reasons why children and cocktail parties do not mix. One less obvious reason is that children struggle to hear and understand speech when multiple people are talking at the same time. Cherry (1953) was not likely thinking about children when he coined the “cocktail party problem” over 60 years ago, referring to the speech perception difficulties individuals often face in social environments with multiple sources of competing sound. Subsequent research has largely focused on trying to understand how adults recognize what one person is saying when other people are talking at the same time (reviewed by Bronkhorst, 2000; McDermott, 2009). However, modern classrooms pose many of the same challenges as a cocktail party, with multiple simultaneous talkers and dynamic listening conditions (Brill et al., 2018). In contrast to the cocktail party, however, failure to recognize speech in a classroom can have important consequences for a child’s educational achievement and social development. These concerns have prompted several laboratories, including ours, to study development of the ability to recognize speech in multisource backgrounds. This article summarizes findings from the smaller number of studies that have examined the cocktail party problem in children, providing evidence that children are at an even greater disadvantage than adults in complex acoustic environments that contain multiple sources of competing sounds.

For much of the school day, children are tasked with listening to their teacher in the context of sounds produced by a range of different sound sources in the classroom. Under these conditions, we would call the teacher’s voice the target and the background sounds would be the maskers. All sounds in the environment, including the target and the maskers, combine in the air before reaching the child’s ears. This combination of acoustic waveforms is often referred to as an *auditory scene*. An example of an auditory scene is illustrated in **Figure 1**, where sounds include the relatively steady noise produced by a projector as well as more dynamic sounds, such as speech produced by classmates who are talking at the same time as their teacher. To hear and understand the teacher, the spectral and temporal characteristics of this mixture of incoming sounds must be accurately represented by the outer ear, middle ear, cochlea, and auditory nerve. This processing is often referred to as *peripheral encoding*. Auditory perception is critically dependent on the peripheral encoding of sound and the fidelity with which this information is transmitted to the brain. Processing within the central auditory system is then needed to identify and group the acoustic waveforms that were generated by the teacher from those that were generated by the other sources (sound source segregation) and then allocate attention to the auditory “object” corresponding to the teacher’s voice while discounting competing sounds (selective auditory attention). Auditory scene analysis also relies on cognitive processes, such as memory, as well as listening experience and linguistic knowledge. Collectively, these processes are often referred to as *auditory scene analysis* (e.g., Bregman, 1990; Darwin and Hukin, 1999).

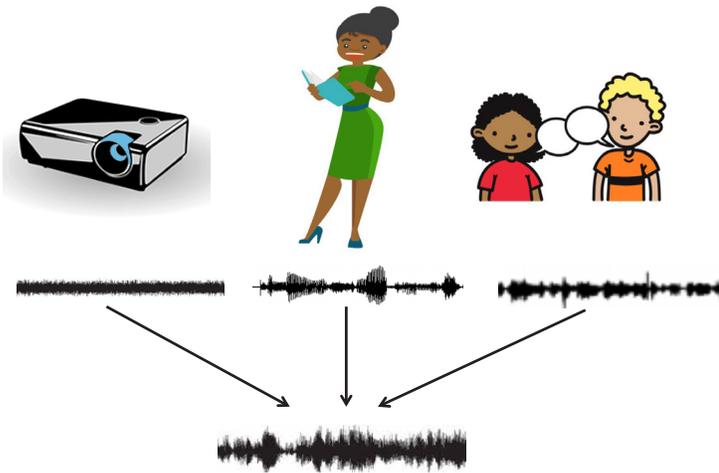


Figure 1. This cartoon illustrates the cocktail party problem in the classroom. In this example, acoustic waveforms are produced by three sources: (1) noise is produced by a computer projector in the classroom; (2) speech is produced by the teacher; and (3) speech is produced by two classmates who are also talking. The fundamental problem is that the acoustic waveforms produced by all three sound sources combine in the air before arriving at the students' ears. To follow the teacher's voice, students must "hear out" and attend to their teacher while disregarding the sounds produced by all other sources.

Immaturity at any stage of processing can impact the extent to which students in the classroom hear and understand the target voice. For example, spectral resolution refers to the ability to resolve the individual frequency components of a complex sound. Degraded spectral resolution is one consequence of congenital hearing loss, specifically sensorineural hearing loss caused by damage to the outer hair cells in the cochlea. This degraded peripheral encoding may reduce audibility of the target speech, making it impossible for adults or children with sensorineural hearing loss to perform auditory scene analysis. Perhaps less obvious, immature central auditory processing could result in the same functional outcome in a child with normal hearing. For example, the perceptual consequence of a failure to selectively attend to the speech stream produced by the teacher, while ignoring classmates' speech, is reduced speech understanding, even when the peripheral encoding of the teacher's speech provides all the cues required for recognition.

Maturation of Peripheral Encoding

Accurate peripheral encoding of speech is clearly a prerequisite for speech recognition. However, sensory representation of the frequency, temporal, and intensity properties of sound does not appear to limit auditory scene analysis during the school-age years. The cochlea begins to function in utero, before the onset of visual functioning (Gottlieb, 1991). Physiological responses to sound provide evidence that the cochlea

is mature by term birth, if not earlier (e.g., Abdala, 2001). Neural transmission through the auditory brainstem appears to be slowed during early infancy, but peripheral encoding of the basic properties of sound approaches the resolution observed for adults by about six months of age (reviewed by Eggermont and Moore, 2012; Vick, 2018).

A competing noise masker can interfere with the peripheral encoding of target speech if the neural excitation produced by the masker overlaps with the neural representation of the target speech. This type of masking can be more severe in children and adults with sensorineural hearing loss than in those with normal hearing. Sensorineural hearing loss is often due to the loss of outer hair cells in the cochlea (reviewed by Moore, 2007). As mentioned above, outer hair cell loss degrades the peripheral encoding of the frequency, intensity, and temporal features of speech, which, in turn, impacts masked speech recognition. Indeed, multiple researchers have demonstrated an association between estimates of peripheral encoding and performance on speech-in-noise tasks for adults with sensorineural hearing loss (e.g., Dubno et al., 1984; Frisina and Frisina, 1997).

Additional evidence that competing noise interferes with the perceptual encoding of speech comes from the results of studies evaluating consonant identification in noise by adults (e.g., Miller and Nicely, 1955; Phatak et al., 2008). Consonant identification is compromised in a systematic way across individuals with normal hearing when competing noise is present, presumably because patterns of excitation produced by the target consonants and masking noise overlap on the basilar membrane (Miller, 1947). In the classroom example shown in **Figure 1**, overlap in excitation patterns between speech produced by the teacher and noise produced by the projector can result in an impoverished neural representation of the teacher's spoken message, although this depends on the relative levels of the two sources and distance to the listener. The term *energetic masking* is often used to describe the perceptual consequences of this phenomenon (reviewed by Brungart, 2005).

Despite mature peripheral encoding, school-age children have more difficulty understanding speech in noise compared with adults. For example, 5- to 7-year-old children require a 3-6 dB more favorable signal-to-noise ratio (SNR) than adults to achieve comparable speech detection, word identification, or sentence recognition performance in a speech-shaped noise masker (e.g., Corbin et al., 2016). Speech-in-noise recognition gradually improves until 9-10 years of age, after which mature performance is generally ob-

served (e.g., Wightman and Kistler, 2005; Nishi et al., 2010). The pronounced difficulties experienced by younger school-age children are somewhat perplexing in light of data indicating that the peripheral encoding of sound matures early in life. It has been posited that these age effects reflect an immature ability to recognize degraded speech (e.g., Eisenberg et al., 2000; Buss et al., 2017). It has also been suggested that children's immature working memory skills also play a role in their speech-in-noise difficulties (McCreery et al., 2017).

Maturation of Auditory Scene Analysis

Children are at a disadvantage relative to adults when listening to speech in competing noise, but the child/adult difference is considerably larger when the maskers are also composed of speech. Hall et al. (2002) compared word recognition for children (5-10 years) and adults tested in each of two maskers: noise filtered to have the same power spectrum as speech (speech-shaped noise; see Multimedia File 1 at acousticstoday.org/leibold-media) and competing speech composed of two people talking at the same time (see Multimedia File 2 at acousticstoday.org/leibold-media). On average, children required a 3 dB more favorable SNR relative to adults to achieve a comparable performance in the noise masker. This disadvantage increased to 8 dB in the two-talker masker. In addition to the relatively large child/adult differences observed in the two-talker masker relative to the noise masker, the ability to recognize masked speech develops at different rates for these two types of maskers (e.g., Corbin et al. 2016). Although adult-like speech recognition in competing noise emerges by 9-10 years of age (e.g., Wightman and Kistler, 2005; Nishi et al., 2010), speech recognition performance in a two-talker speech masker is not adult-like until 13-14 years of age (Corbin et al., 2016). This prolonged time course of development appears to be at least partly due to immature sound segregation and selective attention skills. Recognition of speech produced by the teacher is likely to be limited more by speech produced by other children in the classroom than by noise produced by the projector (see **Figure 1**). The term *informational masking* is often used to refer to this phenomenon (e.g., Brungart, 2005).

An important goal for researchers who study auditory development is to characterize the factors that both facilitate and limit children's ability to perform auditory scene analysis (e.g., Newman et al., 2015; Calandruccio et al., 2016). For listeners of all ages, the perceptual similarity between target and masker speech affects performance in that greater masking is associated with greater perceptual similarity. A common approach to understanding the development of auditory scene

analysis is to measure the extent to which children rely on acoustic voice differences between talkers to segregate target from masker speech (e.g., Flaherty et al., 2018; Leibold et al., 2018). For example, striking effects have been found between conditions in which the target and masker speech are produced by talkers that differ in sex (e.g., a female target talker and a two-male-talker masker) and conditions in which target and masker speech are produced by talkers of the same sex (e.g., a male target talker and a two-male-talker masker). Dramatic improvements in speech intelligibility, as much as 20 percentage points, have been reported in the literature for sex-mismatched relative to sex-matched conditions (e.g., Helfer and Freyman, 2008).

School-age (Wightman and Kistler, 2005; Leibold et al., 2018) and 30-month-old (Newman and Morini, 2017) children also show a robust benefit of a target/masker sex mismatch, but infants younger than 16 months of age do not (Newman and Morini, 2017; Leibold et al., 2018). Leibold et al. (2018), for example, measured speech detection in a two-talker masker in 7- to 13-month-old infants and in adults. Adults performed better when the target word and masker speech were mismatched in sex than when they were matched. In sharp contrast, infants performed similarly in sex-matched and sex-mismatched conditions. The overall pattern of results observed across studies suggest that the ability to take advantage of acoustic voice differences between male and female talkers requires experience with different talkers before the ability emerges sometime between infancy and the preschool years.

Although children as young as 30 months of age benefit from a target/masker sex mismatch, the ability to use more subtle and/or less redundant acoustic voice differences may take longer to develop. Flaherty et al. (2018) tested this hypothesis by examining whether children (5-15 years) and adults benefited from a difference in voice pitch (i.e., fundamental frequency; F0) between target words and a two-talker speech masker, holding other voice characteristics constant. As previously observed for adults (e.g., Darwin et al, 2003), adults and children older than 13 years of age performed substantially better when the target and masker speech differed in F0 than when the F0 of the target and masker speech was matched. This improvement was observed even for the smallest target/masker F0 difference of three semitones. In sharp contrast, younger children (<7 years) did not benefit from even the most extreme F0 difference of nine semitones. Moreover, although 8-12 year olds benefitted from the largest F0 difference, they generally failed to take advantage of more

subtle F0 differences between target and masker speech. These data highlight the importance of auditory experience and maturational effects in learning how to segregate target from masker speech.

In addition to relying on acoustic voice differences between talkers when listening in complex auditory environments, adults with normal hearing take advantage of the differences in signals arriving at the two ears. These differences provide critical information regarding the location of sound sources in space, which, in turn, facilitates segregation of target and masker speech (e.g., Bregman, 1990; Freyman et al., 2001). The binaural benefit associated with separating the target and masker on the horizontal plane is often called spatial release from masking (SRM). In the laboratory, SRM is typically estimated by computing the difference in speech recognition performance between two conditions: the co-located condition, in which the target and masker stimuli are presented from the same location in space, and the spatial separation condition, in which the target and masker stimuli are perceived as originating from different locations on the horizontal plane. For adults with normal hearing, SRM is substantially larger for speech recognition in a masker composed of one or two streams of speech than in a noise masker (reviewed by Bronkhorst, 2000).

Several studies have evaluated SRM in young children and demonstrate a robust benefit of spatially separating the target and masker speech (e.g., Litovsky, 2005; Yuen and Yuan, 2014). Results are mixed, however, regarding the time course of development for SRM. Although Litovsky (2005) observed adult-like SRM in 3-year-old children, other studies have reported a smaller SRM for children compared with adults, a child/adult difference that remains until adolescence (e.g., Yuen and Yuan, 2014; Corbin et al., 2017). In a recent study, Corbin et al. (2017) assessed sentence recognition for children (8-10 years) and adults (18-30 years) tested in a noise masker and in a two-talker masker. Target sentences were always presented from a speaker directly in front of the listener, and the masker was either presented from the front (co-located) or from 90° to the side (separated). Although a comparable SRM was observed between children and adults in the noise masker, the SRM was smaller for children than adults in the two-talker masker. In other words, children benefitted from binaural difference cues less than adults in the speech masker. This is important from a functional perspective because it means that not only are children more detrimentally affected by background speech, but they are

also less able to use spatial cues to overcome the masking associated with speech.

In addition to sound source segregation, auditory scene analysis depends on the ability to allocate and focus attention on the target. Findings from studies using behavioral shadowing procedures provide indirect evidence that selective auditory attention remains immature well into the school-age years (e.g., Doyle, 1973; Wightman and Kistler, 2005). In a typical shadowing task, listeners are asked to repeat speech presented to one ear while ignoring speech or other sounds presented to the opposite ear. Children perform more poorly than adults on these tasks, with age-related improvements observed into the adolescent years (e.g., Doyle, 1973; Wightman and Kistler, 2005). Moreover, children's incorrect responses tend to be intrusions from speech presented to the ear they are supposed to disregard. For example, Wightman and Kistler (2005) asked children (4-16 years) and adults (20-30 years) to attend to target speech presented to the right ear while disregarding masker speech presented to both the right and left ears. Most of the incorrect responses made by adults and children older than 13 years of age were due to confusions with the masker speech that was presented to the same ear as the target speech. In contrast, incorrect responses made by the youngest children (4-5 years) tested were often the result of confusions with the masker speech presented to the opposite ear as the target speech. This result is interpreted as showing that young children do not reliably focus their attention on the target even in the absence of energetic masking.

Although behavioral data suggest that selective auditory attention remains immature throughout most of childhood, a key limitation of existing behavioral paradigms is that we cannot be certain to what a child is or is not attending. Poor performance on a shadowing task might reflect a failure of selective attention to the target but is also consistent with an inability to segregate the two streams of speech (reviewed by Sussman, 2017). This issue is further complicated by the bidirectional relationship between segregation and attention; attention influences the formation of auditory streams (e.g., Shamma et al., 2011). Researchers have begun to disentangle the independent effects of selective auditory attention by measuring auditory event-related brain potentials (ERPs) to both attended and unattended sounds (e.g., Sussman and Steinschneider, 2009; Karns et al., 2015). The pattern of results observed across studies indicates that adult-like ERPs associated with selective auditory attention do not emerge until sometime after 10 years of age, consistent with the time

course of maturation observed in behavioral speech-recognition data and improvements in executive control (reviewed by Crone, 2009).

Role of Linguistic Experience and Knowledge

It has been suggested that the ability to use the information provided by the peripheral auditory system optimally requires years of experience with sound, particularly exposure to spoken language (e.g., Tomblin and Moeller, 2015). In a recent study, Lang et al. (2017) tested a group of 5- to 6-year-old children and found a strong relationship between receptive vocabulary and speech recognition when the masker was two-talker speech masker. As shown in **Figure 2**, children with larger vocabularies were more adept at recognizing sentences presented in a background of two competing talkers than children with more limited vocabularies. Results from previous studies investigating the association between vocabulary and speech recognition in a steady noise masker have been somewhat mixed (e.g., Nittrouer et al., 2013; McCreery et al., 2017). The strong correlation observed by Lang et al. (2016) between vocabulary and speech recognition in a two-talker masker may reflect the greater perceptual and linguistic demands required to segregate and attend to target speech in a speech masker or to the spectrotemporally sparse cues available in dynamic speech maskers.

A second line of evidence that immature language abilities contribute to children's increased difficulty recognizing speech when a few people are talking at the same time comes from studies that have compared children's and adults' ability to recognize speech based on impoverished spectral and/or temporal information (e.g., Eisenberg et al., 2000; Buss et al., 2017). For example, adults are able to recognize band-pass-filtered speech based on a narrower bandwidth than children (e.g., Eisenberg et al., 2000; Mlot et al. 2010). One interpretation for this age effect is that children require more information than adults in order to recognize speech because they have less linguistic experience.

This hypothesis was recently tested by assessing speech recognition in a two-talker masker across a wide age range of children (5-16 years) and adults using speech that was digitally processed using a technique designed to isolate the auditory stream associated with the target speech (Buss et al., 2017). Children and adults showed better performance after the signal processing was applied, indicating that sound source seg-

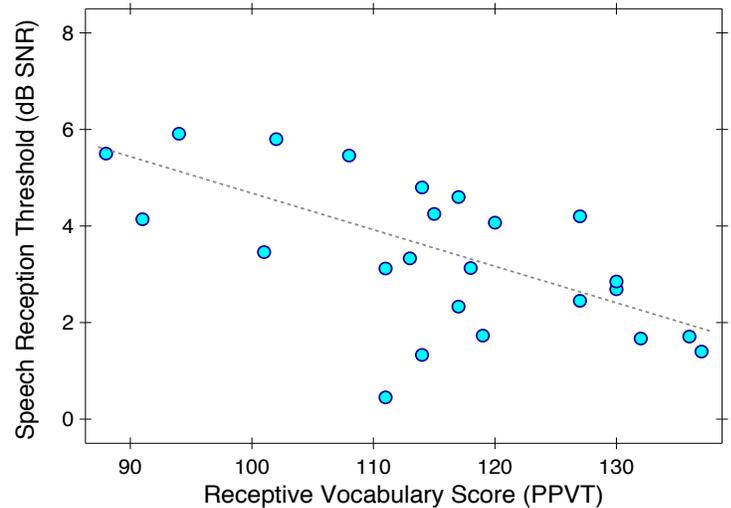


Figure 2. Receptive vocabulary scores and thresholds for sentence recognition in a two-talker masker are shown for 30 young children (5-6 years) tested by Lang et al. (2017). There was a strong association between performance on these two measures ($r = -0.75$; $P < 0.001$), indicating that children with larger vocabularies showed better speech recognition performance in the presence of two competing talkers than children with smaller vocabularies. SNR, signal-to-noise ratio; PPVT, Peabody Picture Vocabulary Test.

regation negatively impacts children's speech recognition in a speech masker. The child/adult difference in performance persisted, however, providing evidence of developmental effects in the ability to reconstruct speech based on sparse speech cues.

Implications

The negative effects of environmental noise on children's speech understanding in the classroom are well documented, leading to the development of a classroom acoustics standard by the Acoustical Society of America (ASA) that was first approved by the American National Standards Institute (ANSI) in 2002 (ANSI S12.60). Although this and subsequent standards recognize the negative effects of environmental noise in the classroom on children's speech understanding, they focus exclusively on noise sources measured in unoccupied classrooms (e.g., heating and ventilation systems, street traffic). The additional sounds typically present in an occupied classroom, such as speech, are not accounted for. As argued by Brill et al. (2018) in an article in *Acoustics Today*, meeting the acoustics standards specified for unoccupied classrooms might not be adequate for ensuring children's speech understanding in occupied classrooms, in which multiple people are often talking at the same time. This is problematic because, as anyone who has spent time in a classroom can attest, children spend most of their days listening and learning with competing speech in the background (e.g., Ambrose et al., 2014; Brill et al., 2018).

Conclusion

Emerging results from investigation into how children listen and learn in multisource environments provide strong evidence that children do not belong at cocktail parties. Despite the more obvious reasons, children lack the extensive linguistic knowledge and the perceptual and cognitive abilities that help adults reconstruct the auditory scene.

Acknowledgments

This work was supported by the Grants R01-DC-011038 and R01-DC-014460 from the National Institute on Deafness and Other Communication Disorders, National Institutes of Health.

References

- Abdala, C. (2001). Maturation of the human cochlear amplifier: Distortion product otoacoustic emission suppression tuning curves recorded at low and high primary tone levels. *The Journal of the Acoustical Society of America* 110, 1465-1476.
- Ambrose, S. E., VanDam, M., and Moeller, M. P. (2014). Linguistic input, electronic media, and communication outcomes of toddlers with hearing loss. *Ear and Hearing* 35, 139-147.
- American National Standards Institute (ANSI). (2002). *S12.60-2002 Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools*. Acoustical Society of America, Melville, NY.
- Bregman, A. S. (1990). *Auditory Scene Analysis: The Perceptual Organization of Sound*. The MIT Press, Cambridge, MA.
- Brill, L. C., Smith, K., and Wang, L. M. (2018). Building a sound future for students – Considering the acoustics in occupied active classrooms. *Acoustics Today* 14(3), 14-22.
- Bronkhorst, A. W. (2000). The cocktail party phenomenon: A review of research on speech intelligibility in multiple-talker conditions. *Acta Acustica united with Acustica* 86, 117-128.
- Brungart, D. S. (2005). Informational and energetic masking effects in multitalker speech perception. In P. Divenyi (Ed.), *Speech Separation by Humans and Machines*. Kluwer Academic Publishers, Boston, MA, pp. 261-267.
- Buss, E., Leibold, L. J., Porter, H. L., and Grose, J. H. (2017). Speech recognition in one- and two-talker maskers in school-age children and adults: Development of perceptual masking and glimpsing. *The Journal of the Acoustical Society of America* 141, 2650-2660.
- Calandruccio, L., Leibold, L. J., and Buss, E. (2016). Linguistic masking release in school-age children and adults. *American Journal of Audiology* 25, 34-40.
- Cherry, E. C. (1953). Some experiments on the recognition of speech, with one and with two ears. *The Journal of the Acoustical Society of America* 25, 975-979.
- Corbin, N. E., Bonino, A. Y., Buss, E., and Leibold, L. J. (2016). Development of open-set word recognition in children: Speech-shaped noise and two-talker speech maskers. *Ear and Hearing* 37, 55-63.
- Corbin, N. E., Buss, E., and Leibold, L. J. (2017). Spatial release from masking in children: Effects of simulated unilateral hearing loss. *Ear and Hearing* 38, 223-235.
- Crone, E. A. (2009). Executive functions in adolescence: Inferences from brain and behavior. *Developmental Science* 12, 825-830.
- Darwin, C. J., Brungart, D. S., and Simpson, B. D. (2003). Effects of fundamental frequency and vocal-tract length changes on attention to one of two simultaneous talkers. *The Journal of the Acoustical Society of America*, 114, 2913-2922.
- Darwin, C. J., and Hukin, R. W. (1999). Auditory objects of attention: The role of interaural time differences. *Journal of Experimental Psychology: Human Perception and Performance* 25, 617-629.
- Doyle, A. B. (1973). Listening to distraction: A developmental study of selective attention. *Journal of Experimental Child Psychology* 15, 100-115.
- Dubno, J. R., Dirks, D. D., and Morgan, D. E. (1984). Effects of age and mild hearing loss on speech recognition in noise. *The Journal of the Acoustical Society of America* 76, 87-96.
- Eggermont, J. J., and Moore, J. K. (2012). Morphological and functional development of the auditory nervous system. In L. A. Werner, R. R. Fay, and A. N. Popper (Eds.), *Human Auditory Development*. Springer-Verlag, New York, pp. 61-106.
- Eisenberg, L. S., Shannon, R. V., Schaefer Martinez, A., Wygonski, J. and Boothroyd, A. (2000). Speech recognition with reduced spectral cues as a function of age. *The Journal of the Acoustical Society of America* 107, 2704-2710.
- Flaherty, M. M., Buss, E., and Leibold, L. J. (2018). Developmental effects in children's ability to benefit from F0 differences between target and masker speech. *Ear and Hearing*. <https://doi.org/10.1097/AUD.0000000000000673>.
- Freyman, R. L., Balakrishnan, U., and Helfer, K. S. (2001). Spatial release from informational masking in speech recognition. *The Journal of the Acoustical Society of America* 109, 2112-2122.
- Frisina, D. R., and Frisina, R. D. (1997). Speech recognition in noise and presbycusis: Relations to possible neural mechanisms. *Hearing Research* 106, 95-104.
- Gottlieb, G. (1991). Epigenetic systems view of human development. *Developmental Psychology* 27, 33-34.
- Hall, J. W., III, Grose, J. H., Buss, E., and Dev, M. B. (2002). Spondee recognition in a two-talker masker and a speech-shaped noise masker in adults and children. *Ear and Hearing* 23, 159-165.
- Helfer, K. S., and Freyman, R. L. (2008). Aging and speech-on-speech masking. *Ear and Hearing* 29, 87-98.
- Karns, C. M., Isbell, E., Giuliano, R. J., and Neville, H. J. (2015). Auditory attention in childhood and adolescence: An event-related potential study of spatial selective attention to one of two simultaneous stories. *Developmental Cognitive Neuroscience* 13, 3-67.
- Lang, H., McCreery, R. W., Leibold, L. J., Buss, E., and Miller, M. K. (2017). Effects of language and cognition on children's masked speech perception. Presented at the 44th Annual Scientific and Technology Meeting of the American Auditory Society, Scottsdale, AZ, March 2-4, 2017, poster #171 - SP12. Available at https://aas.memberclicks.net/assets/2017_posters.pdf. Accessed October 25, 2018.
- Leibold, L. J., Buss, E., and Calandruccio, L. (2018). developmental effects in masking release for speech-in-speech perception due to a target/masker sex mismatch. *Ear and Hearing* 39, 935-945.
- Litovsky, R. Y. (2005). Speech intelligibility and spatial release from masking in young children. *The Journal of the Acoustical Society of America* 117, 3091-3099.
- McCreery, R. W., Spratford, M., Kirby, B., and Brennan, M. (2017). Individual differences in language and working memory affect children's speech recognition in noise. *International Journal of Audiology* 56, 306-315.
- McDermott, J. H. (2009). The cocktail party problem. *Current Biology* 19, R1024-R1027.
- Miller, G. A. (1947). The masking of speech. *Psychological Bulletin* 44, 105-129.
- Miller, G. A., and Nicely, P. E. (1955). An analysis of perceptual confusions among some English consonants. *The Journal of the Acoustical Society of America* 27, 338-352.
- Mlot, S., Buss, E., and Hall, J. W., III. (2010). Spectral integration and bandwidth effects on speech recognition in school-aged children and adults. *Ear and Hearing* 31, 56-62.

Moore, B. C. J. (2007). *Cochlear Hearing Loss: Physiological, Psychological and Technical Issues*. John Wiley & Sons, Hoboken, NJ.

Newman, R. S., and Morini, G. (2017). Effect of the relationship between target and masker sex on infants' recognition of speech. *The Journal of the Acoustical Society of America* 141, EL164-EL169.

Newman, R. S., Morini, G., Ahsan, F., and Kidd, G., Jr. (2015). Linguistically-based informational masking in preschool children. *The Journal of the Acoustical Society of America* 138, EL93-EL98.

Nishi, K., Lewis, D. W., Hoover, B. M., Choi, S., and Stelmachowicz, P. G. (2010). Children's recognition of American English consonants in noise. *The Journal of the Acoustical Society of America* 127, 3177-3188.

Nittrouer, S., Caldwell-Tarr, A., Tarr, E., Lowenstein, J. H., Rice, C., and Moberly, A.C. (2013). Improving speech-in-noise recognition for children with hearing loss: Potential effects of language abilities, binaural summation, and head shadow. *International Journal of Audiology* 52, 513-525.

Phatak, S. A., Lovitt, A., and Allen, J. B. (2008). Consonant confusions in white noise. *The Journal of the Acoustical Society of America* 124, 1220-1233.

Shamma, S. A., Elhilali, M., and Micheyl, C. (2011). Temporal coherence and attention in auditory scene analysis. *Trends in Neurosciences* 34, 114-123.

Sussman, E. S. (2017). Auditory scene analysis: An attention perspective. *Journal of Speech, Language, and Hearing Research* 60, 2989-3000.

Sussman, E., and Steinschneider, M. (2009). Attention effects on auditory scene analysis in children. *Neuropsychologia* 47, 771-785.

Tomblin, J. B., and Moeller, M. P. (2015). An introduction to the outcomes of children with hearing loss study. *Ear and Hearing* 36, 4S-13S.

Vick, J. C. (2018). Advancing toward what will be: Speech development in infancy and early childhood. *Acoustics Today* 14(3), 39-47.

Wightman, F. L., and Kistler, D. J. (2005). Informational masking of speech in children: Effects of ipsilateral and contralateral distracters. *The Journal of the Acoustical Society of America* 118, 3164-3176.

Yuen, K. C., and Yuan, M. (2014). Development of spatial release from masking in Mandarin-speaking children with normal hearing. *Journal of Speech, Language, and Hearing Research* 57, 2005-2023.

BioSketches



Lori Leibold directs the Center for Hearing Research at the Boys Town National Research Hospital, Omaha, NE. She received her BSc in biology from McMaster University, Hamilton, ON, Canada, her MSc in audiology from the University of Western Ontario, London, ON, Canada, and her PhD in hearing science from the University of Washington, Seattle. Her research focuses on auditory development, with a particular interest in understanding how infants and children learn to hear and process target sounds such as speech in the presence of competing background sounds.



Emily Buss is a professor at the University of North Carolina, Chapel Hill, where she serves as chief of auditory research. She received a BA from Swarthmore College, Swarthmore, PA, and a PhD in psychoacoustics from the University of Pennsylvania, Philadelphia. Her research interests include auditory development, the effects of advanced age and hearing loss, auditory prostheses, speech perception, and normative psychoacoustics.



Lauren Calandruccio is an associate professor in the Department of Psychological Sciences, Case Western Reserve University, Cleveland, OH. She received a BA in speech and hearing science and an MA in audiology from Indiana University, Bloomington. After working as a pediatric audiologist, she attended Syracuse University, Syracuse, NY, where she earned her PhD in hearing science. Her research focuses on speech perception, with a focus on noisy or degraded auditory environments, bilingual speech perception, and improving clinical auditory assessment techniques.

INTERNS

An opportunity exists for graduate students and early career acousticians who are members of ASA to serve the Society and gain experience in the publication of a major scientific magazine.

Acoustics Today interns can contribute in a number of different ways, including writing articles for the magazine or website, working with social media to feature AT articles, and many other creative ways. Prospective interns should contact Dr. Popper to explore ideas.

apopper@umd.edu

Heptuna's Contributions to Biosonar

Patrick Moore

Address:

Bioacoustics
National Marine Mammal Foundation
San Diego, California 92152
USA

Email:

Patrick.Moore@nmmf.org

Arthur N. Popper

Address:

Department of Biology
University of Maryland
College Park, Maryland 20742
USA

Email:

apopper@umd.edu

The dolphin Heptuna participated in over 30 studies that helped define what is known about biosonar.

It is not often that it can be said that an animal has a research career that spans four decades or has been a major contributor (and subject) in more than 30 papers in peer-reviewed journals (including many in *The Journal of the Acoustical Society of America*). However, one animal that accomplished this was a remarkable bottlenose dolphin (*Tursiops truncatus*) by the name of Heptuna (**Figure 1**). Indeed, considering the quality of Heptuna's "publications," we contend that were he human and a faculty member at a university, he would easily have been promoted to full professor many years ago.



Figure 1. Heptuna during sound localization studies of Renaud and Popper, ca. 1972.

Heptuna passed away in August 2010 after a career of 40 years in the US Navy. Because Heptuna had such a long and fascinating career and contributed to so much of what we know about marine mammal biosonar, we thought it would be of considerable interest to show the range of studies in which he participated.

Both of the current authors, at one time or another, worked with Heptuna and, like everyone else who worked with the animal, found him to be a bright and effective research subject and, indeed, "collaborator." At the same time, we want to point out that Heptuna, although an exceptional animal, was not unique in having a long and very productive research career; however, he is the focus of this article because both authors worked him and, in many ways, he was truly exceptional.

Early History

Heptuna was collected in midsummer 1970 off the west coast of Florida by US Navy personnel trained in dolphin collection. He was flown to the Marine Air Corps Station Kaneohe Bay in Hawaii that then housed a major US Navy dolphin research facility. At the time of collection, Heptuna was estimated to be about 6 years old, based on his length (close to 2.5 meters, 8 feet) and weight (102 kg, 225 lb). His name came from a multivitamin tablet that, at the time, was a supplement given to all the dolphins at the Naval Undersea Center (NUC).

Heptuna's Basic Training

When Heptuna first joined the Navy, he went through the standard “basic training” procedures used for all Navy dolphins. Exceptionally skilled trainers provided conditioning and acclimation to the Hawaii environment. Heptuna was a fast learner and quickly picked up a fundamental understanding of visual (hand signals) and tonal stimuli needed for specific experiments.

One of the things that made Heptuna such a great experimental animal, leading to his involvement in so many experiments, was that he had a great memory of past experimental procedures and response paradigms. Heptuna also had a willingness to work with the experimenter to figure out what he was supposed to learn. Sometimes, for the less experienced investigator, Heptuna would teach the experimenter.

Heptuna's First Study

After initial training, Heptuna's first research project was a study of sound source localization by University of Hawaii zoology graduate student, Donna McDonald (later Donna McDonald Renaud;¹ **Figure 2**). Donna's mentor, Dr. Arthur Popper, had not worked with dolphins, but he knew a few people at the NUC. He reached out to that group, and they were intrigued by the idea of working with a doctoral student. Donna, Art, and the leadership at the NUC decided that a sound localization study would be of greatest interest because no one had asked if and how well dolphins could localize sound (though it was assumed that they could determine the location of the sound source).

Donna was trained by Ralph Penner, the extraordinary head trainer at the NUC and Heptuna's first trainer. Donna and Ralph first trained Heptuna to swim to a bite bar (a task he would use in many subsequent studies; **Figure 2**), hold his head still, and listen for a sound that came from distant underwater speakers. Heptuna had to discriminate sounds coming from his right or left as the speakers were moved closer together. The goal was to determine the minimal angle between the speakers that could be discriminated (MAA). As soon as Heptuna determined the position of the source, he would swim to a response point on the left or right (**Figure 3**). If he was correct, Donna would reward him with food.

Heptuna quickly picked up the task and for more than two years gave Donna interesting data that examined sound localization abilities for different frequencies and sounds (Re-

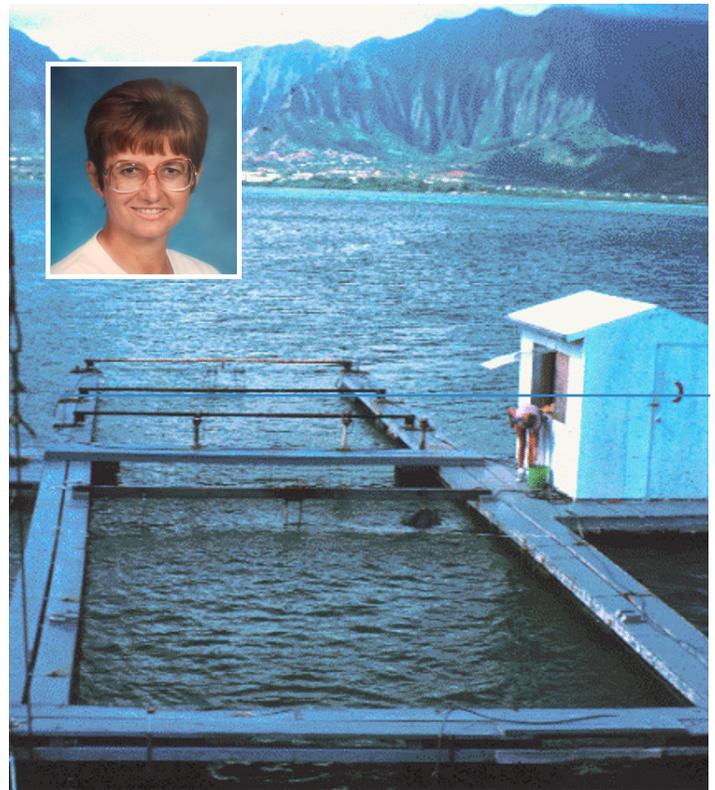


Figure 2. Heptuna's pen for training in Kaneohe Bay, Hawaii. The speakers for the sound localization are at the far bars and the bite bar to which Heptuna held on is in the center of the first cross bar. Equipment is in the shack, and Donna Renaud is seen preparing to throw food to Heptuna. **Inset:** picture of Donna from 1980, kindly provided by her husband Maurice. Donna passed away in 1991.

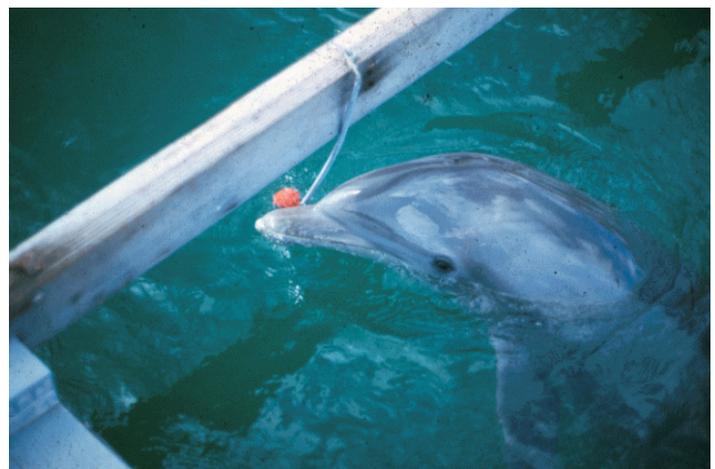


Figure 3. Heptuna hitting the response paddle, ca. 1972. These were the days before many electronics, so the trainer had to see what the animal did and then reward him immediately.

¹This article is dedicated to the memory of Dr. Donna McDonald Renaud (1944-1991), the first investigator to do research with Heptuna.

naud and Popper, 1975). The results showed that Heptuna (and other dolphins) could localize sounds in water almost as well as humans can in air.

Because dolphins live in a three-dimensional acoustic world. Donna decided to ask whether Heptuna could localize sound in the vertical plane. The problem was that the study site was quite shallow and there was no way to put sources above and below the animal. To resolve this, Donna decided that if she could not bring the vertical plane to Heptuna, she would bring Heptuna to the vertical plane. She switched the bite bar to a vertical position and trained Heptuna to do the whole study on his side. As a consequence, the same sound sources that were left and right in the earlier experiments were now above and below the animal's head. Donna found that the MAA for vertical localization was as good as that for horizontal localization, suggesting a remarkably sophisticated localization ability in dolphins.

An Overview of Heptuna's Studies

Shortly after completing the localization study, Heptuna started to “collaborate” with another well-known male dolphin, Sven. Heptuna and Sven began training to detect targets on a “Sky Hook” device, which moved and placed calibrated stainless steel spheres and cylinders underwater at various distances from the echolocating animals. The purpose of the Sky Hook was to determine the maximum distance at which dolphins could detect objects of different sizes and types (Au et al., 1978).

As training continued, Dr. Whitlow Au, a senior scientist at the NUC, attended the sessions conducted by Ralph Penner or Arthur (Earl) Murchison. Whit recorded the animals outgoing echolocation signals (see Au, 2015 for a general history of dolphin biosonar research). These signals were analyzed in an attempt to understand and quantify the echolocation abilities of dolphins. As Whit clearly stated, “In order to better understand the echolocation process and quantify the echolocation sensitivity of odontocetes, it is important to determine the signal-to-noise ratio (SNR) at detection threshold” (Au and Penner, 1981, p. 687). Whit wanted to refine his earlier estimates of the animal's detection threshold based on the transient form of the sonar equation.

Because the earlier thresholds were done in Kaneohe Bay where the background noise was variable and not uniform with respect to frequency, a flat noise floor was needed. Thus, Heptuna was exposed to an added “nearly white-noise source” that was broadcast while he performed the target detection task. The results showed that at the 75% detection

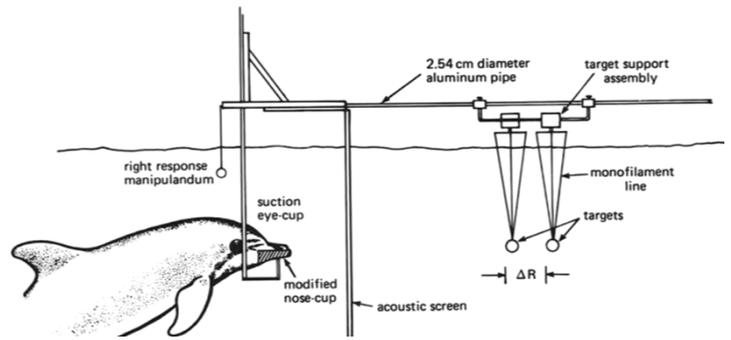


Figure 4. A cartoon of Heptuna's station and the apparatus Earl Murchison used to present the targets (see text for details). ΔR , change in preselected target distances. From Murchison (1980).

threshold, Heptuna's threshold was at a level of 77.3 dB re 1 μ Pa/Hz, whereas it was 74.8 dB for a second dolphin, Ehiku. Whit went on to speculate that this difference may have been due to the abilities of the two animals to distinguish time separation pitch (TSP). In human hearing, TSP is the sensation of a perceived pitch due repetition rate. For dolphins, the concept suggests that the ripples in the frequency domain of echoes provides a cue, an idea that persists in modeling dolphin sonar today (Murchison, 1976; Au, 1988).

Heptuna and Echolocation Studies

Heptuna's biosonar career continued under the tutelage of Earl and Ralph. Earl had begun a long series of experiments on echolocation range resolution of dolphins, and Heptuna was the animal of choice because of his experience. Heptuna faced a new experimental paradigm in this study, requiring him to place his rostrum in a “chin cup” stationing device and echolocate suspended polyurethane foam targets to his left and right (Figure 4). His task was to press a paddle on his right or left corresponding to the closer target. Earl would randomly adjust the targets to one of three preselected ranges (1, 3, or 7 meters). Heptuna was stationed behind an acoustically opaque visual screen so that he could not see or echolocate the targets, and Earl would move one target ever so slightly, moving it a set distance closer or further away in relation to the test range.

Heptuna was the subject for three of Earl's studies of range resolution. Earl found that Heptuna's performance indicated that his range resolution conformed closely to the Weber-Fechner function. In human psychophysics, the law relates to the *perception* of a stimulus; the magnitude of the stimulus when it is just noticeable is a constant ratio (K) of the original stimulus ($\Delta D/D = K$) or conforms to Stevens power law. The results led Earl to speculate how Heptuna's performance would compare with the results of other echolocation experiments. One important observation from Heptuna's results

led Earl to suggest that Heptuna (and by extension other dolphins) had the ability to focus his echolocation attention on a segment of time that encompassed the returning target echo, allowing the dolphin to ignore sounds before and after the echo (Murchison, 1980).

When Earl finished his range resolution studies, Heptuna became available for other research. Whit Au wanted to continue to explore Heptuna's hearing abilities. Because Whit and Patrick Moore had worked together on earlier experiments involving sea lion sound source localization (Moore and Au, 1975), they teamed with Heptuna to better understand his hearing using receiving models from classic sonar acoustics. This began a multiyear research effort to characterize Heptuna's hearing (e.g., Moore and Au, 1983; Au and Moore, 1984; Branstetter et al., 2007).

The first task was to collect masked hearing thresholds at unexplored high frequencies and compute critical ratios and critical bands (Moore and Au, 1983). Armed with these data, it was possible to start a series of experiments to measure Heptuna's horizontal- and vertical-receiving beam patterns (Au and Moore, 1984). This was the first attempt to quantify the receiving beam pattern for an echolocating marine mammal. The investigators used a special pen with a 3.5-meter arc that filled two sides of a 9-square-meter pen. Heptuna stationed at the origin of the arc on a bite plate and was required to remain steady during a trial. Having Heptuna grab the bite plate was easy as this was something he had done in earlier studies using a chin cup. Still, having Heptuna transition to the new 1.5-meter depth of the bite plate proved to be a challenge. Heptuna did not like the aluminum pole used to suspend the bite plate above his head to hold it in position for the horizontal measurements. Wild-born dolphins avoid things over their heads, which is why it is necessary to train them to swim through underwater gates with overhead supports.

Once Heptuna was satisfied that the overhead pole was not going to attack him, the experiment began. The arc in the pen allowed positioning of the signal source about Heptuna's location in 5° increments. For the horizontal beam measurements, two matched noise sources were placed at 20° to the animal's left and right and the investigators could move the signal source. During all of the testing, Heptuna's stationing was monitored by an overhead television camera to ensure he was correctly stationed and unmoving.

After data acquisition for the horizontal beam was finished, Heptuna moved to the vertical beam for measurements. Again, Heptuna proved to be a dolphin of habit. He tried ev-

ery possible way to station on the vertical bite plate just as he had before on the horizontal bite plate except turning on his side. This was perplexing because this was a behavior that Donna McDonald had used in her study (it was found later that he twisted in the opposite direction for Donna). The issue was overcome by slowly rotating the bite plate from the horizontal position to the vertical position over several training sessions and then Heptuna started the vertical measurements. From these data, it was possible to compute Heptuna's directivity index and model a matched pair of receiving transducers that had the same directivity as the animal. Two major observations of the beam patterns were that as the frequency increased, the receiving beam became more narrow (a similar result was found in the bat; Caspers and Müller, 2015) and that the receiving beam was much broader and overlapped with the transmit beam (Au and Moore, 1984).

Heptuna and Controlled Echolocation

Moore's observations over several echolocation experiments found a wide variation in clicks, some with a high source level but with a lower frequency and vice versa. The question became, "does the dolphin have conscious control over click content?" Can he change both the level and frequency of the click as needed to solve an echolocation problem or is it a fixed system so that as the dolphin increases the level, the peak frequency also increases?

Again, Heptuna was chosen to help answer this question. This began a very difficult and long experiment involving echolocation. Heptuna took part in a series of experiments designed to tease out if the dolphin actually had cognitive control over the fine structure of the emitted click. Dr. Ronald Schusterman had already demonstrated tonal control over a dolphin's echolocation click emission. Schusterman et al. (1980) trained a dolphin to echolocate on a target only when a tone stimulus was present and to remain silent when it was not. This experiment started with the attempt to place both Heptuna's echolocation click source level and frequency content under stimulus control while he was actively detecting a target echo.

Heptuna had to learn to station on a bite plate and then place his tail on a tail rest bar behind him, close to his fluke. This stationing procedure was necessary to ensure that Heptuna was stable and aligned with the click-receiving hydrophone, ensuring on-axis sampling of his clicks. Heptuna found this new positioning not at all to his liking. And much like the vertical bite plate with the beam pattern measurements issue mentioned in **Heptuna and Echolocation Studies**, Heptuna

avoided that tail rest pole. He moved his tail up, down, right, and left, always trying to not have that “thing” touch his tail. By systematic and precise reinforcement of small tail movements, however, Heptuna finally touched the device with his tail.

With Heptuna finally positioned correctly, it was possible to start the detection training. Whereas the stationing training took weeks, Heptuna was 100% in detection performance in just one 100-trial session! To capture outgoing clicks, Marion Ceruti, a colleague, and Whit developed a computerized system that could analyze the echolocation click train that Heptuna emitted, computing both the overall peak level and peak frequency of the emitted clicks while doing a real target detection task (Ceruti and Au, 1983).

During a trial, a computer monitored Heptuna’s outgoing clicks and would alert the experimenter if Heptuna met the criterion of either high or low source level or high or low peak frequency and whether the signal was correct. When the computer sounded a high-frequency pure tone, Heptuna would emit loud clicks above the criterion, and when the computer sounded a lower frequency tone, he would keep his clicks below the level criterion. The experimenters also established a frequency criterion, and when the computer sounded a fast-pulsed tone, Heptuna was to keep his peak frequency above a fixed frequency, whereas when the pulses were slow, he kept his peak frequency below a fixed criterion. After intensive training, the experimenters managed to develop stimulus control over Heptuna’s click emissions. As a full demonstration that Heptuna had learned this complex behavior, mixed tones and pulse rates signaled him to produce high-level, low-frequency clicks and vice versa. Heptuna had learned to change his emitted level and peak frequency during an echolocation detection trial and demonstrated conscious control of his echolocation clicks (**Figure 5**; Moore and Pawloski, 1990).

Because Heptuna could produce high source level clicks, above 200 dB re 1 μ Pa (at 1 meter), Ken Norris, one of the great pioneers of dolphin echolocation studies, thought that Heptuna could test the prey-stunning theory that he and Bertel Møhl (see the article about Møhl in *Acoustics Today* by Wahlberg and Au, 2018) had been developing. The hypothesis was that with their very high intensity clicks, dolphins could stun potential prey, making capture much easier. Thus, began a truly exciting experiment involving Heptuna, fish in plastic bags, and suspension devices to hold the bags in front of the animal as he produced very high source level clicks. Bags burst because of bad suspension, sending fresh fish swimming away, with Heptuna giving chase. After many

Voluntary control of adaptive signal

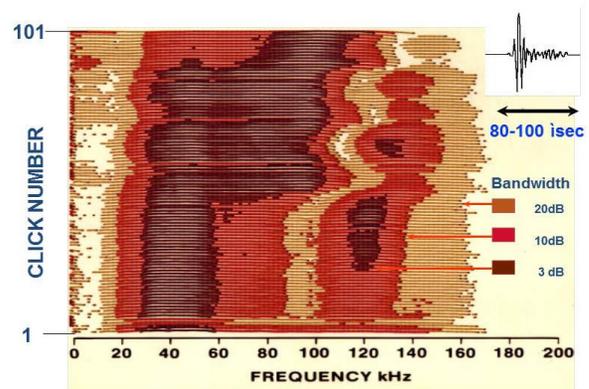


Figure 5. A train of 101 echolocation clicks that Heptuna emitted in the echo detection phase of the experiment. Each horizontal line, starting at the bottom of the figure, is an emitted click. The darker the line colors, the greater the energy across the frequency band. The click train begins with Heptuna emitting narrowband, low-frequency clicks with major energy in the 30- to 60-kHz region. As the click train evolves (around click 12), Heptuna adds energy in the higher frequencies (at 120 kHz,) emitting bimodal energy clicks. The click train develops around click 20 with Heptuna producing very wideband clicks with energy across the frequency spectrum (30 to 110 kHz). The click train ends with Heptuna shifting (around click 85) to clicks with narrowband energy across the 65- to 110-kHz band. This click train lasted just a few seconds. From Moore and Pawloski (1990).

false starts, the bag size, suspension apparatus, and Heptuna were under control. The results did not, however, support the idea of prey stunning by dolphin clicks (Marten et al., 1988).

During this set of experiments, Heptuna had excellent control of his head placement, and Whit wanted to take advantage of the animal’s stationing to refine his vertical emission beam pattern measurements. Heptuna’s positioning was a level of improvement in accuracy over Whit’s first emitted-beam measurements. For this experiment, the control computer would signal Heptuna to echolocate the target (1.3-centimeter-diameter solid steel sphere located 6.4 meters in front of the bite plate) and report whether it was present or absent. Whit used six vertical hydrophones to measure Heptuna’s emitted beam for each click emitted. Whit computed Heptuna’s composite beam pattern over 2,111 beam measurements and showed that the vertical beam was elevated by 5° above the line of Heptuna’s teeth. Whit then calculated the vertical beam to be 15° different from his first measurements. He considered this difference to be attributable to both differences in head anatomy and better control over stationing by Heptuna (Au et al., 1986a).

Heptuna and “Jawphones”

William (Bill) Evans, a graduate student of Dr. Kenneth Norris, used contact hydrophones in suction cups to measure

dolphin emitted signals. Bill's work was groundbreaking echolocation research (Evans, 1967). Using Bill's idea but in reverse, Moore developed the concept of "jawphones" to test dolphin interaural hearing by measuring interaural intensity and time differences. The first pair of jawphones used a Brüel & Kjær (B&K) 8103 miniature hydrophone positioned horizontally along the lower jaw of the animal for maximum efficiency (Figure 6). This position was used because the pathway of sound to the ear in dolphins is through the lower jaw (Brill and Harder, 1991).

Heptuna had no issues with the jawphones because he was trained to wear them as eye cups to occlude his vision for past echolocation experiments. The jawphones were attached to Heptuna's lower jaws, and subsequent thresholds for the pure-tone stimuli were determined. To assess Heptuna's interaural intensity difference threshold, the level of the stimulus was set at a 30-40 dB sensation level (SL). The study used wideband clicks that were similar to dolphin echolocation clicks but that were more suited to the animal's hearing and better represented signals that the animal would naturally encounter. Stimuli were set to a repetition rate corresponding to a target echolocated at 20 meters (40 ms). Using a modified method of constants and a two-alternative forced-choice response paradigm, data were collected for both interaural intensity and time difference thresholds. The results clearly indicated that the dolphin was a highly sophisticated listener and capable of using both time and intensity differences to localize direct and reflected sounds (Moore et al., 1995).

Heptuna Moves to San Diego

In 1992, the Hawaii laboratory was closed and the personnel and animals moved to what is now the Space and Naval Warfare Systems (SPAWAR) Center in San Diego, CA.

Randy Brill, who was then working at SPAWAR, wanted to try to see if there were specific areas of acoustic sensitivity along the lower jaw of the dolphin and other areas around the head. The first thing Randy wanted was to collect thresholds from Heptuna and a second younger animal named Cascade. Using the matched jawphones, it was possible to collect independent thresholds for both the right and left ears of both animals in the background noise of San Diego Bay.

The resulting audiograms for Cascade revealed well-matched hearing in both ears (Brill et al., 2001). However, the results for Heptuna were startling because they showed that Heptuna, now about 33 years old, had hearing loss in both ears, with a more substantial loss in his right ear. Furthermore, Heptuna now had a significant hearing loss above 55 kHz.

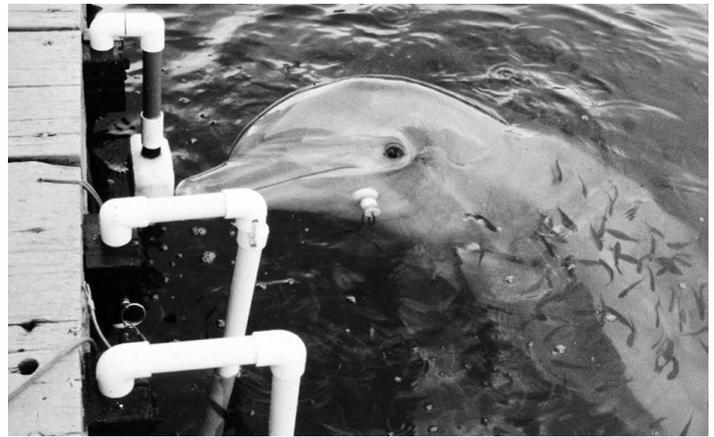


Figure 6. Heptuna wearing "jawphones" during one of Patrick Moore's studies in the early 1990s.

In contrast, when Heptuna was tested at age 26 with the jawphones, his hearing was considered unremarkable because independent thresholds for his ears were closely matched for test frequencies of 4-10 kHz (Moore and Brill, 2001). These data for Heptuna are consistent with the findings of Ridgway and Carder (1993, 1997) showing that dolphins experience age-related hearing loss. Heptuna was another example of a male dolphin losing high-frequency hearing with age, a condition that is similar to presbycusis in humans and that is now known to be common in older dolphins (see the article in *Acoustics Today* by Anderson et al., 2018 about age-related hearing loss in humans).

The results of the free-field thresholds for Cascade at 30, 60, and 90 kHz provided additional support for the use of jawphones as a means to place the sound source in closer proximity to the animal and concentrate the source in a small, localized area. Jawphones have become a tool in the exploration of hearing in dolphins and are used in many experiments conducted at the US Navy Marine Mammal Program and other facilities and in the assessment of hearing in stranded and rehabilitating odontocetes.

Heptuna and Beam Control

Heptuna's hearing loss notwithstanding, the investigators forged ahead to explore the idea that dolphins may control their emitted echolocation beam, allowing them to better detect targets. This involved animals free swimming in the open ocean as they echolocated. Using a new research device that could be carried by the dolphin, the Biosonar Measurement Tool (Houser et al., 2005), it was found that a dolphin could detect echoes from a target before the target entered to animal's main echolocation beam.

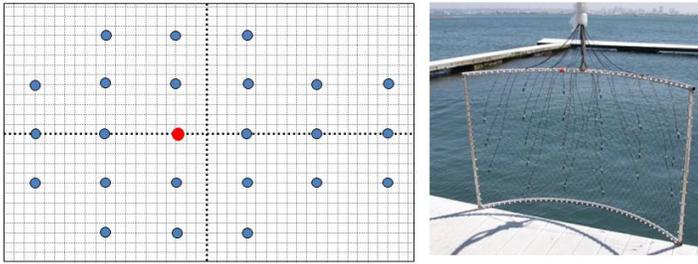


Figure 7. The 24-element hydrophone array used to measure the beam pattern of the dolphin during target detection trials. **Left graphic** shows a planar display of the array arc shown at **right**. **Red star in the center** denotes the P0 hydrophone, which is aligned with the main axis of the dolphin to the target when the target was placed directly in front of the dolphin at P0. From Moore et al. (2008).

This was a behavior that led to an experiment to identify if the dolphin could detect targets off the main beam axis and to quantify their capabilities. To that end, Heptuna was used to explore emitted-beam control. Investigators (Lois Dankiewicz, Dorian Houser, and Moore) devised an experiment to have Heptuna once again station on a bite plate and detect echoes from targets placed at various positions to his right and left. This time, he would be echolocating through a matrix of hydrophones so that the investigators could examine various parameters of each emitted click at each position in the matrix of hydrophones and determine the beam pattern for each click in his emitted click train (**Figure 7**).

Heptuna was asked to detect a water-filled stainless steel sphere and a hollow aluminum cylinder. Heptuna stationed on a bite plate that prevented his ability to move his head during echolocation. He was then asked to echolocate targets as they were placed at various angles to his left and right (Moore et al., 2008). Horizontal and vertical -3 dB beam widths were calculated for each click as well as correlations between click characteristics of peak frequency, center frequency, peak level, and root-mean-square (rms) bandwidth. Differences in the angular detection thresholds for both the sphere and the cylinder to the left and right were relatively equal, and Heptuna could detect the sphere when it was 21° to the right and 26° to the left. His detection threshold for the cylinder was not as good, being only 13° to the right and 19° to the left. The more interesting result became apparent when plotting his composite horizontal and vertical beam patterns. Both were broader than previously measured (Au et al., 1978, 1986b) and varied during target detection. The center part of the beam was also shifted to the right and left when Heptuna was asked to detect targets to the right and left. It was clear that Heptuna's echolocation clicks formed a dynamic

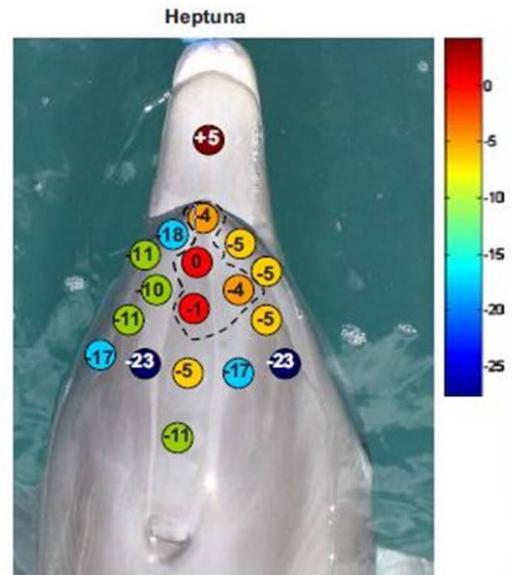


Figure 8. Mean amplitude distribution at various points on Heptuna's head. The colors indicate different intensities (with red being the loudest). The **numbers** are the actual measurements determined at different suction-cup hydrophone locations relative to the loudest sound on the head. **Dashed line**, area of maximum intensity on Heptuna's melon. From Au et al. (2010).

forward-projected but movable beam with complex energy distributions over which the animal had some control.

Heptuna and Contact Hydrophones

Whit Au, pursuing his interest in dolphin echolocation clicks, traveled to San Diego to participate in our ongoing experiments. First, he wanted to determine the location where the echolocation beam axis emerges from the dolphin head and to examine how signals in the acoustic near field relate to signals in the far field. First, investigators (Brian Branstetter, Jim Finneran, and Moore) helped Whit as he placed various hydrophone arrays around the heads of Heptuna and a second dolphin, Bugs (**Figure 8**). Whit collected the clicks from the arrays and computed the position on the melon (a mass in the forehead of all toothed whales that acts as a lens to collimates emitted clicks) where the maximum amplitude of the signals occurred. Whit noted that each dolphin's amplitude gradient about the melon was different, suggesting anatomical differences in both the shape of the forehead and that the sound velocity profile of the animal's melon acted on the emitted signal. Heptuna typically emitted signals with amplitudes higher than those of Bugs by 11-15 dB (Au and Penner, 1981).

Whit also interpreted his results as demonstrating that the animal's emitted click was first shaped internally by the air sacs in its head and then refined by transmission through the melon. Whit suggested that his results supported Norris's

(1968) hypotheses that clicks produced by the phonic lips propagate through a low-velocity core in the melon that positions the emission path almost in the middle of the melon (Au et al., 2010).

An Appreciation

Heptuna's studies described here are really only a "sample" of the work he was engaged in over his 40-year Navy career. The References include many of the research papers that involved Heptuna, and there were also studies that were never published. But the point of this article is that this one animal made substantial contributions to our basic understanding of hearing and echolocation in dolphins. Indeed, Heptuna has become a "legend" in dolphin research. This status likely arose because one of the unique things about Heptuna and, what made him such a valuable animal, was that he learned new tasks remarkably quickly and that he was not easily frustrated. Moreover, he had a really good memory for past training, and he quickly adapted to new tasks based on similar experiences in previous experiments, even many years earlier. And, although it is not quite "scientific" to say it, another thing that promoted Heptuna as an animal (and collaborator) of choice was that he was, from the very beginning in 1971, an easy and friendly animal to work with, something not true of many other dolphins!

Acknowledgments

We thank Dorian Houser and Anthony Hawkins for valuable review of the manuscript. We also thank and acknowledge with great appreciation the many people mentioned in this article for their collaboration and work with Heptuna and the numerous other people who ensured the success of the Navy dolphin research program.

References

Anderson, S., Gordon-Salant, S., and Dubno, J. R. (2018). Hearing and aging effects on speech understanding: Challenges and solutions. *Acoustics Today* 14(4), 10-18.

Au, W. W. L. (2015). History of dolphin biosonar research. *Acoustics Today* 11(4), 10-17.

Au, W. W. L. (1988). Dolphin sonar target detection in noise. *The Journal of the Acoustical Society of America* 84(S1), S133.

Au, W. W. L., Floyd, R. W., and Haun, J. E. (1978). Propagation of Atlantic bottlenose dolphin echolocation signals. *The Journal of the Acoustical Society of America* 64(2), 411-422.

Au, W. W. L., Houser, D. S., Finneran, J. J., Lee, W. J., Talmadge, L. A., and Moore, P. W. (2010). The acoustic field on the forehead of echolocating Atlantic bottlenose dolphins (*Tursiops truncatus*). *The Journal of the Acoustical Society of America* 128, 1426-1434.

Au, W. W. L., and Moore, P. W. B. (1984). Receiving beam patterns and di-

rectivity indices of the Atlantic bottlenose dolphin (*Tursiops truncatus*). *The Journal of the Acoustical Society of America* 75(1), 255-262.

Au, W. W. L., Moore, P. W. B., and Pawloski, D. A. (1986a). The preception of complex echoes by an echolocating dolphin. *The Journal of the Acoustical Society of America* 80(S1), S107.

Au, W. W. L., Moore, P. W. B., and Pawloski, D. (1986b). Echolocation transmitting beam of the Atlantic bottlenose dolphin. *The Journal of the Acoustical Society of America* 80(2), 688-691.

Au, W. W. L., and Penner, R. H. (1981). Target detection in noise by echolocating Atlantic bottlenose dolphins. *The Journal of the Acoustical Society of America* 70(3), 687-693.

Branstetter, B. K., Mercado, E., III, and Au, W. L. (2007). Representing multiple discrimination cues in a computational model of the bottlenose dolphin auditory system. *The Journal of the Acoustical Society of America* 122(4), 2459-2468.

Brill, R. L., and Harder, P. J. (1991). The effects of attenuating returning echolocation signals at the lower jaw of a dolphin (*Tursiops truncatus*). *The Journal of the Acoustical Society of America* 89(6), 2851-2857.

Brill, R. L., Moore, P. W. B., and Dankiewicz, L. A. (2001). Assessment of dolphin (*Tursiops truncatus*) auditory sensitivity and hearing loss using jawphones. *The Journal of the Acoustical Society of America* 109(4), 1717-1722.

Caspers, P., and Müller, R. (2015). Eigenbeam analysis of the diversity in bat biosonar beampatterns. *The Journal of the Acoustical Society of America* 137(3), 1081-1087.

Ceruti, M. G., and Au, W. W. L. (1983). Microprocessor-based system for monitoring a dolphin's echolocation pulse parameters. *The Journal of the Acoustical Society of America* 73(4), 1390-1392.

Evans, W. (1967). Discrimination of different metallic plates by an echolocating delphinid. In R.-G. Busnel (Ed.), *Animal Sonar Systems, Biology and Bionics*. Laboratoire de Physiologie Acoustique, Jouy-en-Josas, France, vol. 1, pp. 363-383.

Houser, D., Martin, S. W., Bauer, E. J., Phillips, M., Herrin, T., Cross, M., Vidal, A., and Moore, P. W. (2005). Echolocation characteristics of free-swimming bottlenose dolphins during object detection and identification. *The Journal of the Acoustical Society of America* 117(4), 2308-2317.

Marten, K., Norris, K. S., Moore, P. W. B., and Englund, K. A. (1988). Loud impulse sounds in odontocete predation and social behavior. In P. E. Nachtigall and P. W. B. Moore (Eds.), *Animal Sonar: Processes and Performance*. Springer US, Boston, MA, pp. 567-579.

Moore, P. W. B., and Au, W. W. L. (1975). Underwater localization of pulsed pure tones by the California sea lion (*Zalophus californianus*). *The Journal of the Acoustical Society of America* 58(3), 721-727.

Moore, P. W. B., and Au, W. W. L. (1983). Critical ratio and bandwidth of the Atlantic bottlenose dolphin (*Tursiops truncatus*). *The Journal of the Acoustical Society of America* 74(S1), S73.

Moore, P. W. B., and Brill, R. L. (2001). Binaural hearing in dolphins. *The Journal of the Acoustical Society of America* 109(5), 2330-2331.

Moore, P. W., Dankiewicz, L. A., and Houser, D. S. (2008). Beamwidth control and angular target detection in an echolocating bottlenose dolphin (*Tursiops truncatus*). *The Journal of the Acoustical Society of America* 124(5), 3324-3332.

Moore, P. W. B., and Pawloski, D. A. (1990). Investigations on the control of echolocation pulses in the dolphin (*Tursiops truncatus*). In J. A. Thomas and R. A. Kastelein (Eds.), *Sensory Abilities of Cetaceans: Laboratory and Field Evidence*. Springer US, Boston, MA, pp. 305-316.

Moore, P. W., Pawloski, D. A., and Dankiewicz, L. (1995). Interaural time and intensity difference thresholds in the bottlenose dolphin (*Tursiops truncatus*). In R. A. Kastelein, J. A. Thomas, and P. E. Nachtigall (Eds.), *Sensory Systems of Aquatic Mammals*. DeSpil Publishers, Woerden, The Netherlands, pp. 11-23.

Murchison, A. E. (1976). Range resolution by an echolocating bottlenosed dolphin (*Tursiops truncatus*). *The Journal of the Acoustical Society of America* 60(S1), S5.

Murchison, A. E. (1980). Detection range and range resolution of echolocating bottlenose porpoise (*Tursiops truncatus*). In R.-G. Busnel and R. H. Penner (Eds.), *Animal Sonar Systems*. Plenum Press, New York, pp. 43-70.

Norris, K. S. (1968). The evolution of acoustic mechanisms in odontocete cetaceans. In E. T. Drake (Ed.), *Evolution and Environment*. Yale University Press, New Haven, CT, pp. 297-324.

Renaud, D. L., and Popper, A. N. (1975). Sound localization by the bottlenose porpoise *Tursiops truncatus*. *Journal of Experimental Biology* 63(3), 569-585.

Ridgway, S. H., and Carder, D. A. (1993). High-frequency hearing loss in old (25+ years old) male dolphins. *The Journal of the Acoustical Society of America* 94(3), 1830.

Ridgway, S. H., and Carder, D. A. (1997). Hearing deficits measured in some *Tursiops truncatus*, and discovery of a deaf/mute dolphin. *The Journal of the Acoustical Society of America*, 101(1), 590-594.

Schusterman, R. J., Kersting, D. A., and Au, W. W. L. (1980). Stimulus control of echolocation pulses in *Tursiops truncatus*. In R.-G. Busnel and J. F. Fish (Eds.), *Animal Sonar Systems*. Springer US, Boston, MA, pp. 981-982.

Wahlberg, M., and Au, W. (2018). Obituary | Bertel Møhl. *Acoustics Today* 14(1), 75.

BioSketches



Patrick W. Moore retired from the US Navy Marine Mammal Program after 42 years of federal service. He was an active scientist as well as a senior manager of the program. Patrick received the Navy Meritorious Civilian Service Award in 1992 and again in 2000 for contributions and leadership in animal psychophysics and neural network models. He is a fellow of the Acoustical Society of America, a charter member of the Society for Marine Mammalogy, and a member of the American Behavior Society and coedited *Animal Sonar: Processes and Performance*. Patrick is currently a senior life scientist at the National Marine Mammal Foundation.



Arthur N. Popper is professor emeritus and research professor at the University of Maryland, College Park, MD. He is also editor of *Acoustics Today* and the Springer Handbook of Auditory Research series. His research focused on hearing by aquatic animals (mostly fishes) as well as on the evolution of vertebrate hearing. For the past 20 years, he has worked on issues related to the effects of anthropogenic sound marine life both in terms of his research and in dealing with policy issues (e.g., development of criteria).

ASA Foundation Fund Launches Major Fundraising Campaign

Campaign for ASA Early Career Leadership

Supporting the Next Generation of Leaders

To further support the next generation of scientists and dedicated leaders in acoustics and the ASA, a new fellowship program, ASA Early Career Leadership Fellows, has been developed. The plan is to award two fellowships annually, each on the order of \$6,000, to support activities that enhance the Fellows' leadership skills and/or potential within their profession and within the ASA, as proposed by the applicants themselves.

LEARN MORE https://acousticalsociety.org/ecl_campaign

Some of the best things in life, and JASA, are free!

Like our collections pages showcasing free JASA articles!

JASA Forum, Review, and Tutorial articles: acousticstoday.org/special-content

JASA Special Issues: acousticstoday.org/JASA-SI

The Remarkable Cochlear Implant and Possibilities for the Next Large Step Forward

Blake S. Wilson

Address:

2410 Wrightwood Avenue
Durham, North Carolina 27705
USA

Also at:

Duke University
Chesterfield Building
701 West Main Street
Room 4122, Suite 410
Durham, North Carolina 27701
USA

Email:

blake.wilson@duke.edu

The modern cochlear implant is an astonishing success; however, room remains for improvement and greater access to this already-marvelous technology.

Introduction

The modern cochlear implant (CI) is a surprising achievement. Many experts in otology and auditory science stated categorically that pervasive and highly synchronous activation of neurons in the auditory nerve with electrical stimuli could not possibly restore useful hearing for deaf or nearly deaf persons. Their argument in essence was “how can one have the hubris to think that the exquisite machinery of the inner ear can be replaced or mimicked with such stimuli?” They had a point!

However, the piece that everyone, or at least most everyone, missed at the beginning and for many years thereafter was the power of the brain to make sense of a sparse and otherwise unnatural input and to make progressively better sense of it over time. In retrospect, the job of designers of CIs was to present just enough information in a clear format at the periphery such that the brain could “take over” and do the rest of the job in perceiving speech and other sounds with adequate accuracy and fidelity. Now we know that the brain is an important part of the prosthesis system, but no one to my knowledge knew that in the early days. The brain “saved us” in producing the wonderful outcomes provided by the present-day CIs.

And indeed, most recipients of those present devices use the telephone routinely, even for conversations with initially unfamiliar persons at the other end and even with unpredictable and changing topics. That is a long trip from total or nearly total deafness!

Now, the CI is widely regarded as one of the great advances in medicine and in engineering. Recently, for example, the development of the modern CI has been recognized by major international awards such as the 2013 Lasker~DeBakey Clinical Medical Research Award and the 2015 Fritz J. and Dolores H. Russ Prize, just to name two among many more.

As of early 2016, more than half a million persons had received a CI on one side or two CIs, with one for each side. That number of recipients exceeds by orders of magnitude the number for any other neural prosthesis (e.g., retinal or vestibular prostheses). Furthermore, the restoration of function with a CI far exceeds the restoration provided by any other neural prosthesis to date.

Of course, the CI is not the first reported substantial restoration of a human sense. The first report, if I am not mistaken, is in the Gospel of Mark in the New Testament (Mark 7:31-37), which describes the restoration of hearing for a deaf man by Jesus. The CI is the first restoration using technology and a medical intervention and is similarly surprising and remarkable.

A Snapshot of the History

The courage of the pioneers made the modern CI possible. They persevered in the face of vociferous criticism, and foremost among them was William F. House, MD, DDS, who with engineer Jack Urban and others developed devices in the late 1960s and early 1970s that could be used by patients in their daily lives outside the laboratory. Additionally, the devices provided an awareness of environmental sounds, were a helpful adjunct to lipreading, and provided limited recognition of speech with the restored hearing alone in rare cases. “Dr. Bill” also developed surgical approaches for placing the CI safely in the cochlea and multiple other surgical innovations, described in his inspiring book (House, 2011). House took most of the arrows from the critics and without his perseverance, the development of the modern CI would have been greatly delayed if not abandoned. He is universally acknowledged as the “Father of Neurotology,” and his towering contributions are lovingly recalled by Laurie S. Eisenberg (2015), who worked closely with him beginning in 1976 and for well over a decade thereafter and stayed in touch with him until his death in 2012.

In my view, five large steps forward led to the devices and treatment modalities we have today. Those steps are

- (1) proof-of-concept demonstrations that a variety of auditory sensations could be elicited with electrical stimulation of the auditory nerve in deaf persons;
- (2) the development of devices that were safe and could function reliably for many years;
- (3) the development of devices that could provide multiple sites of stimulation in the cochlea to take advantage of the tonotopic (frequency) organization of the cochlea and ascending auditory pathways in the brain;
- (4) the discovery and development of processing strategies that utilized the multiple sites far better than before; and
- (5) stimulation in addition to that provided by a unilateral CI, with an additional CI on the opposite side or with acoustic stimulation in conjunction with the unilateral CI.

This list is adapted from a list presented by Wilson (2015).

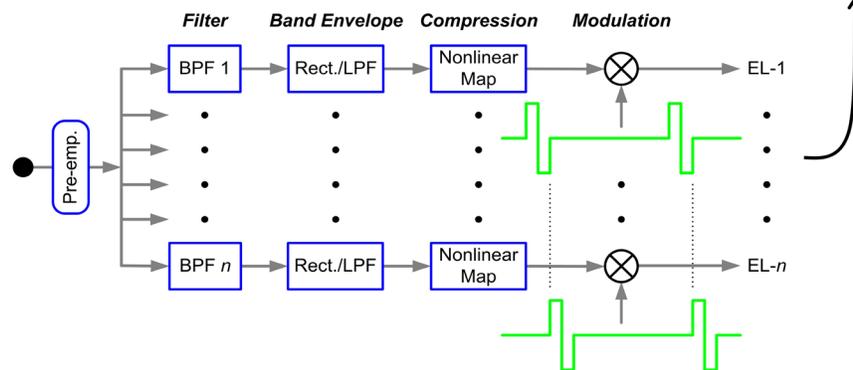
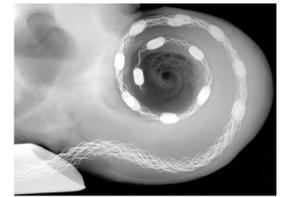


Figure 1. Block diagram of the continuous interleaved sampling (CIS) processing strategy for cochlear implants. **Circles with “x,”** multiplier blocks; **green lines,** carrier waveforms. Band envelopes can be derived in multiple ways and only one way is shown. **Inset:** X-ray image of the implanted cochlea showing the electrode array in the scala tympani. Each channel of processing includes a band-pass filter (BPF); an envelope detector, implemented here with a rectifier (Rect.) followed by a low-pass filter (LPF); a nonlinear mapping function, and the multiplier. The output of each channel is directed to intracochlear electrodes, EL-1 through EL-n, where n is the number of channels. The channel inputs are preceded by a high-pass preemphasis filter (Pre-emp.) to attenuate the strong components at low frequencies in speech, music, and other sounds. Block diagram modified from Wilson et al. (1991), with permission; inset from Hüttenbrink et al. (2002), with permission.

Step 1 was taken by scientist André Djourno and physician Charle Eyriès working together in Paris in 1957 (Seitz, 2002) and step 5 was taken by Christoph von Ilberg in Frankfurt, Joachim Müller in Würzburg, and others in the late 1990s and early 2000s (von Ilberg et al., 1999; Müller et al., 2002; Wilson and Dorman, 2008). Bill House was primarily responsible for step 2, and the first implant operation performed by him was in 1961. Much more information about the history is given by Wilson and Dorman (2008, 2018a), Zeng et al. (2008), and Zeng and Canlon (2015).

A Breakthrough Processing Strategy

Among the five steps, members of the Acoustical Society of America (ASA) may be most interested in step 4, the discovery and development of highly effective processing strategies. A block diagram of the first of those strategies and the progenitor of many of the strategies that followed, is presented in **Figure 1**. The strategy is disarmingly simple and is much simpler than most of its predecessors that included complex analyses of the input sounds to extract and then represent selected features of speech sounds that were judged to be most

important for recognition. Instead, the depicted strategy, continuous interleaved sampling (CIS; Wilson et al., 1991), makes no assumptions about how speech is produced or perceived and simply strives to represent the input in a way that will utilize most or all of the perceptual ranges of electrically evoked hearing as clearly as possible.

As shown, the strategy includes multiple channels of sound processing whose outputs are directed to the different electrodes in an array of electrodes implanted in the scala tympani (ST), one of three fluid-filled chambers along the length of the cochlea (see X-ray inset in **Figure 1**, which shows an electrode array in the ST). The channels differ only in the frequency range for the band-pass filter. The channel outputs with high center frequencies for the filters are directed to electrodes at the basal end of the cochlea, which is most sensitive to high-frequency sounds in normal hearing (the tonotopic organization mentioned in **A Snapshot of the History**), and the channel outputs with lower center frequencies are directed to electrodes toward the other (apical) end of the cochlea, which in normal hearing is most sensitive to sounds at lower frequencies.

The span of the frequencies across the band-pass filters typically is from 300 Hz or lower to 6 kHz or higher, and the distribution of frequencies is logarithmic, like the distribution of frequency sensitivities along the length of the cochlea in normal hearing. In each channel, the varying energy in the band-pass filter is sensed with an envelope detector, and then the output of the detector is “mapped” onto the narrow dynamic range of electrically evoked hearing (5-20 dB for pulses vs. 90 dB or more for normal hearing) using a logarithmic or power-law transformation. The envelope detector can be as simple as a low-pass filter followed by a rectifier (full wave or half wave) or as complex as the envelope output of a Hilbert Transform. Both are effective. The compressed envelope signal from the nonlinear mapping function modulates a carrier of balanced biphasic pulses for each of the channels to represent the energy variations in the input. Those modulated pulse trains are directed to the intracochlear electrodes as previously described. Implant users are sensitive to both place of stimulation in the cochlea or auditory nerve and the rate or frequency of stimulation at each place (Simmons et al., 1965).

Present-day implants include 12-24 intracochlear electrodes; some users can rank all of their electrodes according to pitch, and most users can rank at least a substantial subset of the electrodes when the electrodes are stimulated separately and one at a time. (Note, however, that no more than eight elec-

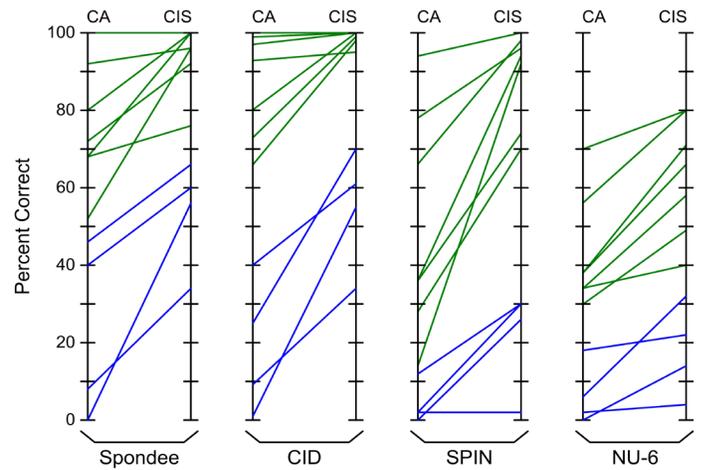


Figure 2. Results from initial comparisons of the compressed analog (CA) and CIS processing strategies. **Green lines**, scores for subjects selected for their exceptionally high levels of performance with the CA strategy; **blue lines**, scores for subjects selected for their more typical levels of performance with that strategy. The tests included recognition of two-syllable words (Spondee); the Central Institute for the Deaf (CID) everyday sentences; sentences from the Speech-in-Noise test (SPIN) but here without the added noise; and the Northwestern University list six of monosyllabic words (NU-6). From Wilson and Dorman (2018a), with permission.

trodes may be effective in a multichannel context, at least for ST implants and the current processing strategies; see Wilson and Dorman, 2008.) Also, users typically perceive increases in pitch with increases in the rate or frequency of stimulation, or the frequency of modulation for modulated pulse trains, at each electrode up to about 300 pulses/s or 300 Hz but with no increases in pitch with further increases in rate or frequency (e.g., Zeng, 2002). For that reason, the cutoff of the low-pass filter in each of the processing channels usually is set at 200-400 Hz to include most or all of the range over which different frequencies in the modulation waveforms can be perceived as different pitches. Fortuitously, the 400-Hz choice also includes the full range of the fundamental frequencies in voiced speech for men, women, and children. The pulse rate for each channel is the same across channels and is usually set at four times the cutoff frequencies (which also are uniform across channels) to minimize ambiguities in the perception of the envelope (modulation) signals that can occur at lower rates (Busby et al., 1993; Wilson et al., 1997).

A further aspect of the processing is to address the effects of the highly conductive fluid in the ST (the perilymph) and the relatively distant placements of the intracochlear electrodes from their neural targets (generally thought to be the spiral ganglion cells in the cochlea). The high conductivity and the

distance combine to produce broad spreads of the excitation fields from each electrode along the length of the cochlea (length constant of about 10 mm or greater compared with the ~35-mm length of the human cochlea). Also, the fields from each electrode overlap strongly with the fields from other electrodes. The aspect of processing is to present the pulses across channels and their associated electrodes in a sequence rather than simultaneously. The nonsimultaneous or “interleaved” stimulation eliminates direct summation of electric fields from the different electrodes that otherwise would sharply degrade the perceptual independence of the channels and electrodes. CIS gets its name from the continuous (and fixed rate) sampling of the mapped envelope signals by interleaved pulses across the channels.

The overall approach is to utilize the perceptual space fully and to present the information in ways that will preserve the independence of the channels and minimize perceptual distortions as much as possible. Of course, in retrospect, this approach also allowed the brain to work its magic. Once we designers “got out of the way” in presenting a relatively clear and unfettered signal rather than doing anything more or more complicated, the brain could take over and do the rest.

Some of the first results from comparisons of CIS with the best strategy in clinical use at the time are presented in **Figure 2**. Results from four tests are shown and range in difficulty from easy to extremely difficult for speech presented in otherwise quiet conditions. Each subject had had at least one year of daily experience with their clinical device and processing strategy, the Ineraid™ CI and the “compressed analog” (CA) strategy, respectively, but no more than several hours of experience with CIS before the tests. (The CA strategy presented compressed analog signals simultaneously to each of four intracochlear electrodes and is described further in Wilson, 2015.) The **green lines** in **Figure 2** show the results for a first set of subjects selected for high performance with the CA strategy (data from Wilson et al., 1991), which was fully representative of the best performances that had been obtained with CIs as of the time of testing. The **blue lines** in **Figure 2** show the results for a second set of subjects who were selected for their more typical levels of performance (data from Wilson et al., 1992). The scores for all tests and subjects demonstrated an immediate and highly significant improvement with CIS compared with the alternative strategy.

Not surprisingly, the subjects were thrilled along with us by this outcome. One of the subjects said, for example, “Now you’ve got it!” and another slapped the table in front of him

and said, “Hot damn, I want to take this one home with me!” All three major manufacturers of CIs (which had more than 99% of the market share) implemented CIS in new versions of their products in record times for medical devices after the results from the first set of subjects were published (Wilson et al., 1991), and CIS became available for widespread clinical use within just a few years thereafter. Thus, the subjects got their wish and the CI users who followed them benefitted as well.

Many other strategies were developed after CIS, but most were based on it (Fayad et al., 2008; Zeng and Canlon, 2015; Zeng, 2017). CIS is still used today and remains as the principal “gold standard” against which newer and potentially beneficial strategies are compared. Much more information about CIS and the strategies that followed it is presented in recent reviews (Wilson and Dorman, 2008, 2012; Zeng et al., 2008). Additionally, most of the prior strategies are described in Tyler et al. (1989) and Wilson (2004, 2015).

Performance of Unilateral Cochlear Implants

The performance for speech reception in otherwise quiet conditions is seen in **Figure 3**, which shows results from two large studies conducted approximately 15 years apart. In **Figure 3**, the **blue circles and lines** show the results from a study conducted by Helms et al. (1997) in the mid-1990s and the **green circles and lines** show the results from tests with patients who were implanted from 2011 to mid-2014 (data courtesy of René Gifford at the Vanderbilt University Medical Center [VUMC]). For both studies, the subjects were postlingually (after the acquisition of language in childhood with normal or nearly normal hearing) deafened adults, and the tests included recognition of sentences and monosyllabic words. The words were comparable in difficulty between the studies, but the low-context Arizona Biomedical (AzBio) sentences used in the VUMC study were more difficult than the high-context Hochmair-Schultz-Moser (HSM) sentences used in the Helms et al. (1997) study. Measures were made at the indicated times after the initial fitting of the device, and the means and standard error of the means (SEMs) of the scores are shown in **Figure 3**. Details about the subjects and tests are presented in Wilson et al. (2016).

The results demonstrate (1) high levels of speech reception for high-context sentences; (2) lower levels for low-context sentences; (3) improvements in the scores for all tests with increasing time out to 3-12 months depending on the test; (4) a complete overlapping of scores at every common test

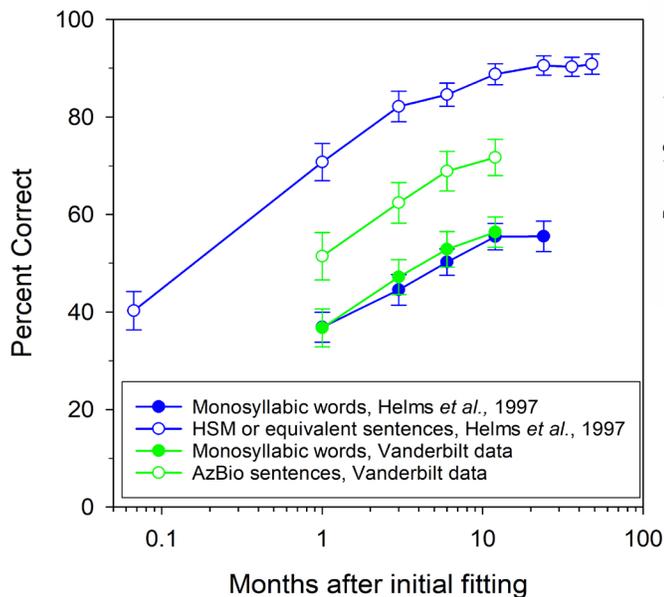


Figure 3. Means and SEMs for recognition of monosyllabic words (**solid circles**) and sentences (**open circles**) by implant subjects. The sentences included the AzBio sentences (**green circles and lines**) and the Hochmair-Schultz-Moser (HSM) sentences in German or their equivalents in other languages (**blue circles and lines**). See text for additional details about the tests and sources of data. From Wilson and Dorman (2018b), with permission.

interval for the two monosyllabic word tests; and (5) lower scores for the word tests than for the sentence tests.

The improvements over time indicate a principal role of the brain in determining outcomes with CIs. In particular, the time course of the improvements is consistent with changes in brain function in adapting to a novel input (Moore and Shannon, 2009) but not consistent with changes at the periphery such as reductions in electrode impedances that occur during the first days, not months, of implant use. The brain makes sense of the input initially and makes progressively better sense of it over time, out to 3-12 months and perhaps even beyond 12 months. (Note that the acute comparisons in **Figure 2** did not capture the improvements over time that might have resulted with substitution of the new processing strategy on a long-term basis; also see Tyler et al., 1986.)

The results from the monosyllabic word tests also indicate that the performance of unilateral CIs has not changed much, if at all, since the early 1990s, when the new processing strategies became available for clinical use (also see Wilson, 2015, for additional data in this regard). These tests are particularly good fiducial markers because the scores for the individual subjects do not encounter ceiling or floor effects for any of the modern CIs and processing strategies tested to date.

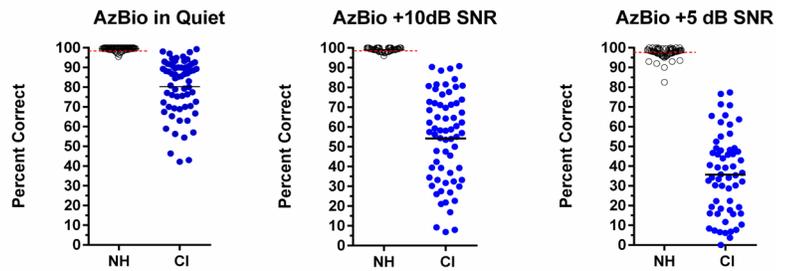


Figure 4. Recognition by subjects with normal hearing (NH; **black circles**) and CI (**blue circles**) subjects of AzBio sentences presented in an otherwise quiet condition (**left**) or in competition with environmental noise at the speech-to-noise ratios (SNRs) of +10 dB (**center**) and +5 dB (**right**). Horizontal lines, means of the scores for each test and set of subjects. From Wilson and Dorman (2018b), with permission; data courtesy of Dr. René Gifford.

An additional aspect of performance with the present-day unilateral CIs is seen in **Figure 4**, which shows the effects of noise interference on performance. These data also are from VUMC and again kindly provided by Dr. Gifford. The subjects include 82 adults with normal hearing (NH) and 60 adult users of unilateral CIs from the same corpus mentioned previously or implanted later at the VUMC. The AzBio sentences were used and were presented in an otherwise quiet condition (**Figure 4, left**) or in competition with environmental noise at the speech-to-noise ratios (SNRs) of +10 (**Figure 4, center**) and +5 dB (**Figure 4, right**). Scores for the individual subjects are shown along with the mean scores indicated by the horizontal lines.

The scores for the NH subjects are at or near 100% correct for the quiet and +10 dB conditions and above 80% correct for the +5 dB condition. In contrast, scores for the CI subjects are much lower for all conditions and do not overlap the NH scores for the +10 and +5 dB conditions. Thus, the present-day unilateral CIs do not provide NH, especially in adverse acoustic conditions such as the ones shown and such as in typically noisy restaurants or workplaces. However, the CIs do provide highly useful hearing in relatively quiet (and reverberation-free) conditions, as shown by the data in **Figure 4, left**, and by the sentence scores in **Figure 3**.

Adjunctive Stimulation

Although the performance of unilateral CIs has been relatively constant for the past 2+ decades, another way has been found to increase performance and that is to present stimuli in addition to the stimuli presented by a unilateral CI. As noted in **A Snapshot of the History**, this additional (or adjunctive) stimulation can be provided with a second CI on the

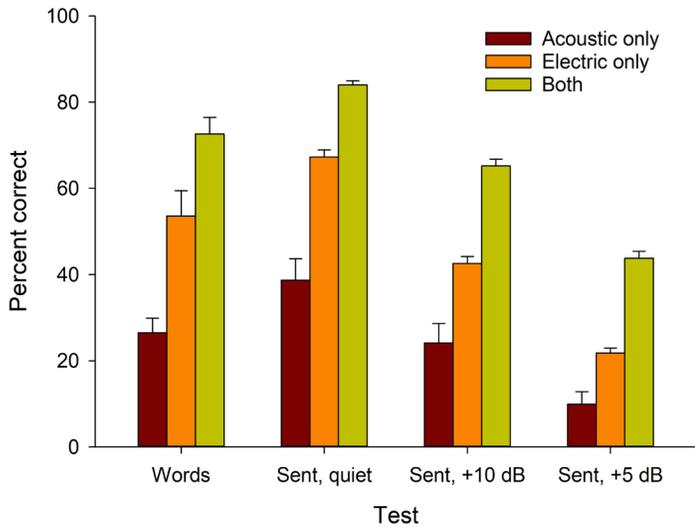


Figure 5. Means and SEMs of scores for the recognition of monosyllabic words (Words), AzBio sentences presented in an otherwise quiet condition (Sent, quiet), and the sentences presented in competition with speech-babble noise at the SNRs of +10 dB (Sent, +10 dB) and +5 dB (Sent, +5 dB). Measures were made with acoustical stimulation of the ear with residual hearing for each of the 15 subjects; electrical stimulation with the CI on the opposite side; and combined electric and acoustic stimulation. Data from Dorman et al. (2008).

opposite side or with acoustic stimulation, the latter for persons with useful residual hearing on either side or both sides.

The effects of the additional stimulation can be large, as seen in **Figure 5**, which shows the effects of combined electric and acoustic stimulation (combined EAS; also called “hybrid” or “bimodal” stimulation). The data are from Dorman et al. (2008). They tested 15 subjects who had a full insertion of a CI on one side; residual hearing at low frequencies on the opposite side; and 5 months to 7 years of experience with the CI and 5 or more years of experience with a hearing aid. The tests included recognition of monosyllabic words and the AzBio sentences with acoustic stimulation of the one ear only with the hearing aid, electric stimulation of the opposite ear only with the CI, and combined EAS. As in **Figure 4**, the sentences were presented in an otherwise quiet condition and in competition with noise (4-talker babble noise) at the SNRs of +10 and +5 dB.

Means and SEMs of the scores are presented in **Figure 5** and demonstrate the large benefits of the combination for all tests. Compared with electric stimulation only, the combination produces a jump up in the recognition of monosyllabic

words from 54 to 73% correct and a 2-fold increase in the recognition of the sentences in noise at the SNR of +5 dB. Thus, the barrier of ~55% correct for recognition of monosyllabic words by experienced users of unilateral CIs (**Figure 3**) can be broken, and recognition of speech in noise can be increased with combined EAS. Excellent results also have been obtained with bilateral electrical stimulation, as shown for example in Müller et al. (2002).

In broad terms, both combined EAS and bilateral CIs can improve speech reception substantially. Also, combined EAS can improve music reception and bilateral CIs can enable sound localization abilities. Furthermore, the brain can integrate the seemingly disparate inputs from electric and acoustic stimulation, or the inputs from the two sides from bilateral electrical stimulation, into unitary percepts that for speech are more intelligible, often far more intelligible, than either input alone. Step 5 was a major step forward.

Step 6?

In my view, the greatest opportunities for the next large step forward are

- (1) increasing access worldwide to the marvelous technology that already has been developed and proven to be safe and highly beneficial;
- (2) improving the performance of unilateral CIs, which is the only option for many patients and prospective patients and is the foundation of the adjunctive stimulation treatments; and
- (3) broadening the eligibility and indications for CIs and the adjunctive treatments, perhaps to include the many millions of people worldwide who suffer from disabling hearing loss in their sixth decade and beyond (a condition called “presbycusis”).

Any of these advances would be a worthy step 6.

Increasing Access

As mentioned in the **Introduction**, slightly more than half a million people worldwide have received a CI or bilateral CIs to date. In contrast, approximately 57 million people worldwide have a severe or worse hearing loss in the better hearing ear (Wilson et al., 2017). Most of these people could benefit from a CI. Additionally, manyfold more, with somewhat better hearing on the worse side or with substantially better hearing on the opposite side, could benefit greatly from combined EAS. A conservative estimate of the number of persons who could benefit from a CI or the adjunctive stimulation treatments is around 60 million and the actual number is probably very much higher. Taking the conser-

vative estimate, approximately 1% of the people who could benefit from a CI has received one.

I think a population health perspective would be helpful in increasing the access; progress already has been made along these lines (Zeng, 2017). Access is limited by the cost of the device but also by the availability of trained medical personnel; the infrastructure for healthcare in a region or country; awareness of the benefits of the CI at the policy levels such as the Ministries of Finance and Ministries of Health; the cost of surgery and follow-up care; additional costs associated with the care for patients in remote regions far from tertiary-care hospitals; battery expenses; the cost for manufacturers in meeting regulatory requirements; the cost for manufacturers in supporting clinics; the cost of marketing where needed; and the cost of at least minimal profits to sustain manufacturing enterprises. Access might be increased by viewing it as a multifaceted problem that includes all of these factors and not just the cost of the device, although that is certainly important (Emmett et al., 2015).

Efforts are underway by Ingeborg J. Hochmair and me and by Fan-Gang Zeng and others to increase access. We know that even under the present conditions, CIs are cost effective or highly cost effective in high- and middle-income countries and are cost effective or approaching cost effectiveness in some of the lower income countries with improving economies (Emmett et al., 2015, 2016; Saunders et al., 2015). However, much more could be done to increase access—especially in the middle- and low-income countries—so that “All may hear,” as Bill House put it years ago, and as was the motto for the House Ear Institute in Los Angeles (founded by Bill’s half brother Howard) before its demise in 2013 (Shannon, 2015).

Improving Unilateral Cochlear Implants

As seen in **Figure 3** and as noted by Lim et al. (2017) and Zeng (2017), the performance of unilateral CIs has been relatively static since the mid-1990s despite many well-conceived efforts to improve them and despite (1) multiple relaxations in the candidacy criteria for cochlear implantation; (2) increases in the number of stimulus sites in the cochlea; and (3) the advent of multiple new devices and processing strategies. Presumably, today’s recipients have healthier cochleas and certainly a higher number of good processing options than the recipients of the mid-1990s. In the mid-1990s, the candidacy criteria were akin to “can you hear a jet engine 3 meters away from you?” and, if not, you could be a candidate. Today, persons with substantial residual hearing, and even persons with a severe or

worse loss in hearing one side but normal or nearly normal hearing on the other side, can be candidates for receiving a CI.

These efforts and differences did not move the needle in the clockwise direction. New approaches are obviously needed, and some of the possibilities are presented by Wilson (2015, 2018), Zeng (2017), and Wilson and Dorman (2018b); one of those possibilities is to pay more attention to the “hearing brain” in designs and applications of CIs.

Better performance with unilateral CIs is important because not all patients or prospective patients have access to, or could benefit from, the adjunctive stimulation treatments. In particular, not all patients have enough residual hearing in either ear to benefit from combined EAS (Dorman et al., 2015), even with the relaxations in the candidacy criteria, and not all patients have access to bilateral CIs due to restrictions in insurance coverage or national health policies. Furthermore, the performance of the unilateral CI is the foundation of the adjunctive treatments and an increase in performance for unilateral CIs would be expected to boost the performance of the adjunctive treatments as well.

Broadening Eligibility and Indications

Even a slight further relaxation in the candidacy criteria, based on data, would increase substantially the number of persons who could benefit from a CI. Evidence for a broadening of eligibility is available today (Gifford et al., 2010; Wilson, 2012).

An immensely large population of persons who would be included as candidates with the slight relaxation are the sufferers of presbycusis, which is a socially isolating and otherwise debilitating condition. There are more than 10 million people in the United States alone who have this affliction, and the numbers in the United States and worldwide are growing exponentially with the ongoing increases in and aging of the world’s populations. A hearing aid often is not effective for presbycusis sufferers because most of them have good or even normal hearing at low frequencies (below about 1.5 kHz) but poor or extremely poor hearing at the higher frequencies (Dubno et al., 2013). The amplification provided by a hearing aid is generally not needed at the low frequencies and is generally not effective (or only marginally effective) at the high frequencies because little remains that can be stimulated acoustically there. A better treatment is needed. Possibly, a shallowly and gently inserted CI could provide a “light tonotopic touch” at the basal (high-frequency) end of the cochlea to complement the low-frequency hearing that already exists for this stunningly large population of potential beneficiaries.

Coda

Although the present-day CIs are wonderful, considerable room remains for improvement and for greater access to the technology that has already been developed.

The modern CI is a shared triumph of engineering, medicine, and neuroscience, among other disciplines. Indeed, many members of our spectacular ASA have contributed mightily in making a seemingly impossible feat possible (see **Box**) and, in retrospect, the brave first steps and cooperation among the disciplines were essential in producing the devices we have today.

Contributions by Members of the Acoustical Society of America

Members of the ASA contributed mightily to the development of the modern CI. Two examples among many are that the citations for 14 Fellows of the ASA have been for contributions to the development and that 556 research articles and 95 letters that include the keywords “cochlear implant” have been published in *The Journal of the Acoustical Society of America* as of September 2018. Additionally, Fellows of the ASA have served as the Chair or Cochair or both for 14 of the 19 biennial “Conferences on Implantable Auditory Prostheses” conducted to date or scheduled for 2019. These conferences are the preeminent research conferences in the field; in all, 18 Fellows have participated or will participate as the Chair or Cochair. Interestingly, the citations for nine of these Fellows were not for the development and that speaks to the multidisciplinary nature of the effort.

In thinking back on the history of the CI, I am reminded of the development of aircraft. At the outset, many experts stated categorically that flight with a heavier-than-air machine was impossible. The pioneers proved that the naysayers were wrong. Later, much later, the DC-3 came along. It is a classic engineering design that remained in widespread use for decades and is still in use today. It transformed air travel and transportation, like the modern CI transformed otology and the lives of the great majority of its users. The DC-3 was surpassed, of course, with substantial investments of resources, high expertise, and unwavering confidence and diligence. I expect the same will happen for the CI.

References

- Busby, P. A., Tong, Y. C., and Clark, G. M. (1993). The perception of temporal modulations by cochlear implant patients. *The Journal of the Acoustical Society of America* 94(1), 124-131.
- Dorman, M. F., Cook, S., Spahr, A., Zhang, T., Loiseau, L., Schramm, D., Whittingham, J., and Gifford, R. (2015). Factors constraining the benefit to speech understanding of combining information from low-frequency hearing and a cochlear implant. *Hearing Research* 322, 107-111.
- Dorman, M. F., Gifford, R. H., Spahr, A. J., and McKarns, S. A. (2008). The benefits of combining acoustic and electric stimulation for the recognition of speech, voice and melodies. *Audiology & Neurotology* 13(2), 105-112.
- Dubno, J. R., Eckert, M. A., Lee, F. S., Matthews, L. J., and Schmiedt, R. A. (2013). Classifying human audiometric phenotypes of age-related hearing loss from animal models. *Journal of the Association for Research in Otolaryngology* 14(5), 687-701.
- Eisenberg, L. S. (2015). The contributions of William F. House to the field of implantable auditory devices. *Hearing Research* 322, 52-66.
- Emmett, S. D., Tucci, D. L., Bento, R. F., Garcia, J. M., Juman, S., Chiossone-Kerdel, J. A., Liu, T. J., De Muñoz, P. C., Ullauri, A., Letort, J. J., and Mansilla, T. (2016). Moving beyond GDP: Cost effectiveness of cochlear implantation and deaf education in Latin America. *Otology & Neurotology* 37(8), 1040-1048.
- Emmett, S. D., Tucci, D. L., Smith, M., Macharia, I. M., Ndegwa, S. N., Nakku, D., Kaitesi, M. B., Ibekwe, T. S., Mulwafu, W., Gong, W., and Francis, H. W. (2015). GDP matters: Cost effectiveness of cochlear implantation and deaf education in sub-Saharan Africa. *Otology & Neurotology* 36(8), 1357-1365.
- Fayad, J. N., Otto, S. R., Shannon, R. V., and Brackman, D. E. (2008). Cochlear and brainstem auditory prostheses “Neural interface for hearing restoration: Cochlear and brain stem implants.” *Proceedings of the IEEE* 96, 1085-1095.
- Gifford, R. H., Dorman, M. F., Shallop, J. K., and Sydlowski, S. A. (2010). Evidence for the expansion of adult cochlear implant candidacy. *Ear & Hearing* 31(2), 186-194.
- Helms, J., Müller, J., Schön, F., Moser, L., Arnold, W., Janssen, T., Ramsden, R., von Ilberg, C., Kiefer, J., Pfennigdorf, T., and Gstöttner, W. (1997). Evaluation of performance with the COMBI 40 cochlear implant in adults: A multicentric clinical study. *ORL Journal of Oto-Rhino-Laryngology and Its Related Specialties* 59(1), 23-35.
- House, W. F. (2011). *The Struggles of a Medical Innovator: Cochlear Implants and Other Ear Surgeries: A Memoir by William F. House, D.D.S., M.D.* CreateSpace Independent Publishing, Charleston, SC.
- Hüttenbrink, K. B., Zahnert, T., Jolly, C., and Hofmann, G. (2002). Movements of cochlear implant electrodes inside the cochlea during insertion: An x-ray microscopy study. *Otology & Neurotology* 23(2), 187-191.
- Lim, H. H., Adams, M. E., Nelson, P. B., and Oxenham, A. J. (2017). Restoring hearing with neural prostheses: Current status and future directions. In K. W. Horch and D. R. Kipke (Eds.), *Neuroprosthetics: Theory and Practice, Second Edition*. World Scientific Publishing, Singapore, pp. 668-709.
- Moore, D. R., and Shannon, R. V. (2009). Beyond cochlear implants: Awakening the deafened brain. *Nature Neuroscience* 12(6), 686-691.
- Müller, J., Schön, F., and Helms J. (2002). Speech understanding in quiet and noise in bilateral users of the MED-EL COMBI 40/40+ cochlear implant system. *Ear & Hearing* 23(3), 198-206.
- Saunders, J. E., Barrs, D. M., Gong, W., Wilson, B. S., Mojica, K., and Tucci, D. L. (2015). Cost effectiveness of childhood cochlear implantation and deaf education in Nicaragua: A disability adjusted life year model. *Otology & Neurotology* 36(8), 1349-1356.

- Seitz, P. R. (2002). French origins of the cochlear implant. *Cochlear Implants International* 3(2), 77-86.
- Shannon, R. V. (2015). Auditory implant research at the House Ear Institute 1989-2013. *Hearing Research* 322, 57-66.
- Simmons, F. B., Epley, J. M., Lummis, R. C., Guttman, N., Frishkopf, L. S., Harmon, L. D., and Zwicker, E. (1965). Auditory nerve: Electrical stimulation in man. *Science* 148(3666), 104-106.
- Tyler, R. S., Moore, B. C., and Kuk, F. K. (1989). Performance of some of the better cochlear-implant patients. *Journal of Speech & Hearing Research* 32(4), 887-911.
- Tyler, R. S., Preece, J. P., Lansing, C. R., Otto, S. R., and Gantz, B. J. (1986). Previous experience as a confounding factor in comparing cochlear-implant processing schemes. *Journal of Speech & Hearing Research* 29(2), 282-287.
- von Ilberg, C., Kiefer, J., Tillein, J., Pfenningdorff, T., Hartmann, R., Stürzebecher, E., and Klinke, R. (1999). Electric-acoustic stimulation of the auditory system: New technology for severe hearing loss. *ORL Journal of Oto-Rhino-Laryngology & Its Related Specialties* 61(6), 334-340.
- Wilson, B. S. (2004). Engineering design of cochlear implants. In F. G. Zeng, A. N. Popper, and R. R. Fay (Eds.), *Cochlear Implants: Auditory Prostheses and Electric Hearing*. Springer-Verlag, New York, pp. 14-52.
- Wilson, B. S. (2012). Treatments for partial deafness using combined electric and acoustic stimulation of the auditory system. *Journal of Hearing Science* 2(2), 19-32.
- Wilson, B. S. (2015). Getting a decent (but sparse) signal to the brain for users of cochlear implants. *Hearing Research* 322, 24-38.
- Wilson, B. S. (2018). The cochlear implant and possibilities for narrowing the remaining gaps between prosthetic and normal hearing. *World Journal of Otorhinolaryngology - Head and Neck Surgery* 3(4), 200-210.
- Wilson, B. S., and Dorman, M. F. (2008). Cochlear implants: A remarkable past and a brilliant future. *Hearing Research* 242(1-2), 3-21.
- Wilson, B. S., and Dorman, M. F. (2012). Signal processing strategies for cochlear implants. In M. J. Ruckenstein (Ed.), *Cochlear Implants and Other Implantable Hearing Devices*. Plural Publishing, San Diego, CA, pp. 51-84.
- Wilson, B. S., and Dorman, M. F. (2018a). A brief history of the cochlear implant and related treatments. In A. Rezzi, E. Krames, and H. Peckham (Eds.), *Neuromodulation: A Comprehensive Handbook*, 2nd ed. Elsevier, Amsterdam, pp. 1197-1207.
- Wilson, B. S., and Dorman, M. F. (2018b). Stimulation for the return of hearing. In A. Rezzi, E. Krames, and H. Peckham (Eds.), *Neuromodulation: A Comprehensive Handbook*, 2nd ed. Elsevier, Amsterdam, pp. 1209-1220.
- Wilson, B. S., Dorman, M. F., Gifford, R. H., and McAlpine, D. (2016). Cochlear implant design considerations. In N. M. Young and K. Iler Kirk (Eds.), *Pediatric Cochlear Implantation: Learning and the Brain*. Springer-Verlag, New York, pp. 3-23.
- Wilson, B. S., Finley, C. C., Lawson, D. T., Wolford, R. D., Eddington, D. K., and Rabinowitz, W. M. (1991). Better speech recognition with cochlear implants. *Nature* 352(6332), 236-238.
- Wilson, B. S., Finley, C. C., Lawson, D. T., and Zerbi, M. (1997). Temporal representations with cochlear implants. *American Journal of Otology* 18(6 Suppl.), S30-S34.
- Wilson, B. S., Lawson, D. T., Zerbi, M., and Finley, C. C. (1992). Speech processors for auditory prostheses: Completion of the "poor performance" series. *Twelfth Quarterly Progress Report, NIH Project N01-DC-9-2401*, Neural Prosthesis Program, National Institute on Deafness and Other Communication Disorders, National Institutes of Health, Bethesda, MD.
- Wilson, B. S., Tucci, D. L., Merson, M. H., and O'Donoghue, G. M. (2017). Global hearing health care: New findings and perspectives. *Lancet* 390(10111), 2503-2515.
- Zeng, F. G. (2002). Temporal pitch in electric hearing. *Hearing Research* 174(1-2), 101-106.
- Zeng, F. G. (2017). Challenges in improving cochlear implant performance and accessibility. *IEEE Transactions on Biomedical Engineering* 64(8), 1662-1664.
- Zeng, F. G., and Canlon, B. (2015). Recognizing the journey and celebrating the achievement of cochlear implants. *Hearing Research* 322, 1-3.
- Zeng, F. G., Rebscher, S., Harrison, W., Sun, X., and Feng, H. (2008). Cochlear implants: System design, integration, and evaluation. *IEEE Reviews in Biomedical Engineering* 1, 115-142.

BioSketch



Blake Wilson is a fellow of the Acoustical Society of America (ASA) and a proud recipient of the ASA Helmholtz-Rayleigh Interdisciplinary Silver Medal in Psychological and Physiological Acoustics, Speech Communication, and Signal Processing in Acoustics, in his case “for

contributions to the development and adoption of cochlear implants.” He also is the recipient of other major awards including the 2013 Lasker Award (along with two others) and the 2015 Russ Prize (along with four others). He is a member of the National Academy of Engineering and of the faculties at Duke University, Durham, NC; The University of Texas at Dallas; and Warwick University, Coventry, UK.

Be Heard!

Students, get more involved with the ASA through our student council at: asastudentcouncil.org/get-involved

ASA

Recent Acoustical Society of America Awards and Prizes

Starting with this issue, *Acoustics Today* will be publishing the names of the recipients of the various awards and prizes given out by the Acoustical Society of America. After the recipients are approved by the Executive Council of the Society at each semiannual meeting, their names will be published in the next issue of *Acoustics Today*.

Congratulations to the following recipients of Acoustical Society of America medals, awards, prizes, and fellowships, many of whom will be formally recognized at the spring 2019 meeting in Louisville, KY. For more information on these accolades, please see acousticstoday.org/asa-awards, acousticalsociety.org/prizes, and acousticstoday.org/fellowships.

2019 Gold Medal

William Cavanaugh
Cavanaugh Tocci Associates

*2019 Helmholtz-Rayleigh
Interdisciplinary Silver Medal in
Psychological and Physiological
Acoustics, Speech Communication,
and Architectural Acoustics*
Barbara Shinn-Cunningham
Carnegie Mellon University

2019 Distinguished Service Citation
David Feit
Acoustical Society of America

2019 R. Bruce Lindsay Award
Adam Maxwell
University of Washington

*2019 Medwin Prize in
Acoustical Oceanography*
Chen-Fen Huang
National Taiwan University

*2019 William and Christine Hartmann
Prize in Auditory Neuroscience*
Glenis Long
City University of New York

2018 Silver Medal in Engineering Acoustics
Thomas B. Gabrielson
Pennsylvania State University

2018 James E. West Fellowship
Dillan Villavisanis
Johns Hopkins University

*2018 Leo and Gabriella Beranek
Scholarship in Architectural Acoustics
and Noise Control*
Kieren Smith
University of Nebraska – Lincoln

*2018 Raymond H. Stetson Scholarship in
Phonetics and Speech Science*
Heather Campbell Kabakoff
New York University
Nicholas Monto
University of Connecticut

*2018 Frank and Virginia Winker
Memorial Scholarship for Graduate
Study in Acoustics*
Caleb Goates
Brigham Young University

Congratulations also to the following members who were elected Fellows in the Acoustical Society of America at the fall 2018 meeting (acoustic.link/ASA-Fellows).

- **Megan S. Ballard** (University of Texas at Austin) for contributions to shallow water propagation and geoacoustic inversion
- **Lori J. Leibold** (Boys Town National Research Hospital) for contributions to our understanding of auditory development
- **Robert W. Pyle** (Harvard University) for contributions to the understanding of the acoustics of brass musical instruments
- **Woojae Seong** (Seoul National University) for contributions to geoacoustic inversion and ocean signal processing
- **Rajka Smiljanic** (University of Texas at Austin) for contributions to cross-language speech acoustics and perception
- **Edward J. Walsh** for contributions to auditory physiology, animal bioacoustics, and public policy

Ask an Acoustician: Kent L. Gee

Kent L. Gee

Address:

Department of Physics and Astronomy
Brigham Young University
N243 ESC
Provo, Utah 84602
USA

Email:

kentgee@byu.edu

Micheal L. Dent

Address:

Department of Psychology
University at Buffalo
State University of New York (SUNY)
B76 Park Hall
Buffalo, New York 14260
USA

Email:

mdent@buffalo.edu



Meet Kent L. Gee

In this “Ask an Acoustician” column, we hear from Kent L. Gee, a professor in physics and astronomy at Brigham Young University (BYU), Provo, UT. If you go to the Acoustical Society of America (ASA) meetings, you have likely seen Kent around. He was awarded the prestigious R. Bruce Lindsay Award in 2010 and became a fellow of the Society in 2015. He currently serves as editor of the *Proceedings of Meetings on Acoustics (POMA)* and is on the Membership Committee. He has developed demonstration shows

for the physical acoustics summer school, advised his local BYU ASA student chapter for more than a decade, has brought dozens of students to the ASA meetings, and has organized numerous ASA sessions. Kent is active in the Education in Acoustics, Noise, and Physical Acoustics Technical Committees of the ASA. So if you think you know him, you probably do! I will let Kent tell you the rest.

A Conversation with Kent Gee, in His Words

Tell us about your work.

My research primarily involves characterizing high-amplitude sound sources and fields. With students and colleagues, I have been able to make unique measurements of sources like military jet aircraft, rockets, and explosions (e.g., Gee et al., 2008, 2013, 2016a,b). Along the way, we’ve developed new signal analysis techniques for both linear and nonlinear acoustics. Whenever possible, I also try to publish in acoustics education (e.g., Gee, 2011). For those interested, nearly all my journal articles and conference publications are found at acousticstoday.org/gee-pubs.

Describe your career path.

When I arrived at BYU as a freshman, I had a plan: major in physics and become a high-school physics teacher and track coach. That plan lasted about one week because I rapidly became disillusioned with the large-lecture format and over-enthusiastic students to whom I simply didn’t relate. But, after a year of general education classes and a two-year religious mission, I found my way back to physics. After another year of studies, I became disenchanted again. I was doing well in my classes, but I didn’t feel excited about many of the topics, at least not enough to want to continue as a “general” physics major. I began to explore various emphases for an applied physics major and soon discovered that acoustics was an option. In my junior year, I began to do research with Scott Sommerfeldt and took a graduate course in acoustics. Although I was underprepared and struggled in the course, I discovered that I absolutely loved the material. That rapidly led to my taking two more graduate courses, obtaining an internship at the NASA Glenn Research Center, taking on additional research projects, fast tracking a master’s degree at BYU, and then pursuing a PhD in acoustics at Pennsylvania State University, University Park, under Vic Sparrow. Along the way, I found that my passion for

acoustics only increased the more I learned. Remarkably, after receiving my doctorate, I found my way back to BYU as a faculty member and I've been here ever since.

What is a typical day for you?

My typical day at work involves juggling teaching, scholarship, mentoring, and service. I currently teach a graduate course on acoustical measurement methods and 550 students in 2 sections of a general education course in physical science. Scholarship and mentorship are intertwined because I work with several graduate and undergraduate students on projects related to military jet noise, rocket noise, sonic booms, vector intensity measurements, and machine learning. Service involves committee work (like chairing the Awards Committee in my department), outreach activities at schools, reviewing journal manuscripts, and being editor of *POMA*. Those that have met me, and perhaps those that haven't, know of my particular passion for the possibilities of *POMA* as a publication. Alliteration aside, I believe *POMA* has an important role to play in the long-term growth and health of the Society, and I work hard to address issues that arise and to expand its global reach and visibility.

How do you feel when experiments/projects do not work out the way you expected them to?

I admit that when a project doesn't work out the way I envisioned because of data anomalies, misunderstandings, or poor experimental design, I tend to dwell on these failures for a long time. Sometimes, the dwelling will be productive and lead to other project possibilities, but when it just can't be furthered, it's tough. On the other hand, when we've discovered something that we didn't anticipate leads to whole new ways of thinking, all the down moments melt away in the excitement of breaking science!

Do you feel like you have solved the work-life balance problem? Was it always this way?

Not in the least. Every day is a battle to decide where and how I can do the most good. I have an amazing wife and 5 wonderful children, ages 9-16, who are growing up too quickly. I have a job that I am passionate about and that allows me to influence the next generation of educators and technological leaders. I have opportunities to serve in the community and at my church. At the end of the day, I try to prioritize by what is most important and most urgent and let other things fall away. I just hope I'm getting better at it.

What makes you a good acoustician?

In all seriousness, this question is probably best asked of someone else. But, any good I have accomplished probably

comes from three things. First, I have strived to be an effective student mentor by learning and employing mentoring principles (Gee and Popper, 2017). I have been blessed to work with remarkable students. I help them navigate the discovery and writing processes and add insights along the way. Second, I have found connections between seemingly disparate research areas of my research and leveraged them for greater understanding and applicability. For example, as a new faculty member, I was able to combine my prior experiences with vector intensity and military jet noise to investigate near-field energy-based measurements of rocket plumes. This study led to improved source models of rockets and a new method for calculating vector intensity from microphone probes that is being applied to a variety of problems, including infrasound from wind turbines. The hardware required for those infrasound measurements was recently developed for space vehicle launches and is now being refined to make high-fidelity recordings of quiet sonic booms in adverse weather conditions. Seeking connections has led to unexpected opportunities for learning. Third, a lesson my father taught me was that hard work and determination can often compensate for lack of natural ability. I hope to always apply that lesson. Perseverance and passion do seem to go a long way.

How do you handle rejection?

Not very well, I'm afraid. I tend to stew and lose sleep over these things. But I'm getting a little better with time, I think. One thing that helps is focusing on the other great things in my life when a grant or project doesn't get funded. So, while it's hard to balance all the things I listed above, it actually helps to balance the ups and downs very naturally.

What are you proudest of in your career?

I am proudest of my students and their accomplishments, both in research and in the classroom. Seeing the good they're doing in the world is no small victory for me.

What is the biggest mistake you've ever made?

Niels Bohr purportedly said, "An expert is a man who has made all the mistakes which can be made in a very narrow field" (available at en.wikiquote.org/wiki/Niels_Bohr). Regrettably, I have not yet achieved expert status in any area of research, teaching, or mentoring, but I'll share one experience from which I've learned. Before my first military jet aircraft measurement at BYU, I programmed the input range of the data-acquisition system in terms of maximum expected voltage instead of expected acoustic pressure. The moment the F-16 engine was fired up, many channels clipped because the hardware input range was set too low. Yet, I wasn't able to figure out the problem until shortly after the measurement was over. I was able to pull

physical insights from several channels that didn't clip, but it was an incredibly stressful experience that taught me to double and triple check hardware configurations before getting into the field. Thankfully, my colleagues were gracious about the mistake and I am able still count them as friends and collaborators today.

What advice do you have for budding acousticians?

I have benefited immensely from exceptional mentoring. Try to be affiliated with people who care at least as much about who you are becoming as about what you are learning and then work as hard as possible to learn from them. The acoustics community is full of this kind of researchers and professionals. Conversely, when faced with those who neither have time nor concern for your progress, keep moving forward. Perseverance and passion!

Have you ever experienced imposter syndrome?

How did you deal with that if so?

In late 2009, I found out that I was going to receive the ASA Lindsay Award. I honestly had a hard time doing much of anything for a few weeks after that because I felt extraordinarily inadequate. Somehow, a lot of smart people had been fooled into thinking I had done something special, and there

was no way that I could sustain that charade. Over time, the feeling of academic paralysis was gradually replaced with a determination to at least try to live up to what others thought I was capable of. Although imposter syndrome doesn't go away, I have learned to recognize it and use it as motivation.

What do you want to accomplish within the next 10 years or before retirement?

I just want to make a difference, whether connecting nonlinearities in jet and rocket noise to human annoyance, developing improved vector measurement methods, or mentoring the next generation of students who will go on to do great things.

References

- Gee, K. L. (2011). The Rubens tube. *Proceedings of Meetings on Acoustics* 8, 025003.
- Gee, K. L., Neilsen, T. B., Downing, J. M., James, M. M., McKinley, R. L., McKinley, R. C., and Wall, A. T. (2013). Near-field shock formation in noise propagation from a high-power jet aircraft. *The Journal of the Acoustical Society of America* 133, EL88-EL93.
- Gee, K. L., Neilsen, T. B., Wall, A. T., Downing, J. M., James, M. M., and McKinley, R. L. (2016a). Propagation of crackle containing noise from military jet aircraft. *Noise Control Engineering Journal* 64, 1-12.
- Gee, K. L., and Popper, A. N. (2017). Improving academic mentoring relationships and environments. *Acoustics Today* 13(3), 27-35.
- Gee, K. L., Sparrow, V. W., James, M. M., Downing, J. M., Hobbs, C. M., Gabrielson, T. B., and Atchley, A. A. (2008). The role of nonlinear effects in the propagation of noise from high-power jet aircraft. *The Journal of the Acoustical Society of America* 123, 4082-4093.
- Gee, K. L., Whiting E. B., Neilsen, T. B., James, M. M., and Salton, A. R. (2016b). Development of a near-field intensity measurement capability for static rocket firings. *Transactions of the Japan Society for Aeronautical and Space Sciences, Aerospace Technology Japan* 14(ists30), Po_2_9-Po_2_15.



**BECOME
AN ASA STANDARDS MEMBER**

MAKE YOUR VOICE HEARD

Help shape the standards that influence your business and bottom line

**National ANSI-Accredited
Standards Committees:**

- S1 Acoustics
- S2 Mechanical Vibration and Shock
- S3 Bioacoustics
 - S3/SC1 Animal Bioacoustics
- S12 Noise

**International ANSI-Accredited
U.S. Technical Advisory Groups:**

- ISO/TC 43 Acoustics
- ISO/TC 43/SC1 Noise
- ISO/TC 43/SC3 Underwater acoustics
- ISO/TC 108 Mechanical vibration, shock and condition monitoring and its subcommittees
- IEC/TC 29 Electroacoustics

Nancy Blair-DeLeon, Standards Manager
Acoustical Society of America Standards Secretariat
asastds@acousticalsociety.org acousticalsociety.org/standards



**Become
Involved!**

**Would you like to become
more involved with the ASA?**

Visit acousticalsociety.org/
volunteer to learn more about
the Society's technical and
administrative committees, and
submit a form to express your
interest in volunteering!

Scientists with Hearing Loss Changing Perspectives in STEMM

Henry J. Adler

Address:
Center for Hearing and Deafness
University at Buffalo
The State University of New York
(SUNY)
Buffalo, New York 14214
USA

Email:
henryadl@buffalo.edu

J. Tilak Ratnanather

Address:
Department of Biomedical Engineering
Johns Hopkins University
Baltimore, Maryland 21218
USA

Email:
tilak@cis.jhu.edu

Peter S. Steyger

Address:
Oregon Hearing Research Center
Oregon Health & Science University
Portland, Oregon 97239
USA

Email:
steygerp@ohsu.edu

Brad N. Buran

Address:
Oregon Hearing Research Center
Oregon Health & Science University
Portland, Oregon 97239
USA

Email:
buran@ohsu.edu

Despite extensive recruitment, minorities remain underrepresented in science, technology, engineering, mathematics, and medicine (STEMM). However, decades of research suggest that diversity yields tangible benefits. Indeed, it is not surprising that teams consisting of individuals with diverse expertise are better at solving problems. However, there are drawbacks to socially diverse teams, such as increased discomfort, lack of trust, and poorer communication. Yet, these are offset by the increased creativity of these teams as they work harder to resolve these issues (Phillips, 2014). Although gender and race typically come to mind when thinking about diversity, people with disabilities also bring unique perspectives and challenges to academic research (think about Stephen Hawking as the most notable example). This is particularly true when they work in a field related to their disability. Here, we briefly introduce four deaf or hard-of-hearing (D/HH) scientists involved in auditory research: Henry J. Adler, J. Tilak Ratnanather, Peter S. Steyger, and Brad N. Buran, the authors of this article. The first three have been in the field since the late 1980s while the fourth has just become an independent investigator. Our purpose is to relay to readers our experiences as D/HH researchers in auditory neuroscience.

More than 80 scientists with hearing loss have conducted auditory science studies in recent years. They include researchers, clinicians, and past trainees worldwide, spanning diverse backgrounds, including gender and ethnicity, and academic interests ranging from audiology to psychoacoustics to molecular biology (Adler et al., 2017). Many have published in *The Journal of the Acoustical Society of America (JASA)*, and recently, Erick Gallun was elected a Fellow of the ASA. Recently, approximately 20 D/HH investigators gathered (see **Figure 1**) at the annual meeting of the Association for Research in Otolaryngology (ARO) that has, in our consensus opinion, set the benchmark for accessibility at scientific conferences.

The perspective of scientists who are D/HH provides novel insights into understanding auditory perception, hearing loss, and restoring auditory functionality. Their identities as D/HH individuals are diverse, and their ability to hear ranges from moderate to profound hearing loss. Likewise, their strategies to overcome spoken language barriers range from writing back and forth (including email or text messaging) to real-time captioning to assistive listening devices to sign language to Cued Speech.

Henry J. Adler

Born with profound hearing loss, I was diagnosed at 11 months of age and have since worn hearing aids. I attended the Lexington School for the Deaf in Jackson Heights, New York, NY, and then was mainstreamed (from 4th grade) into public schools (including the Bronx High School of Science) in New York City. When I was at Lexington, its policy forbade any sign language, and listening and spoken language (LSL) was my primary mode of communication. At Harvard College,



Figure 1. Several of the 80+ scientists with hearing loss discussing their participation at the 2017 Association for Research in Otolaryngology meeting. **Standing front to back:** Patrick Raphael, Daniel Tward*, Brenda Farrell*^, Robert Raphael^, and Tilak Ratnanather^. **Seated, clockwise from left:** Erica Hegland, Kelsey Anbuhl, Patricia Stahn, Valluri “Bob” Rao, Oluwaseun Ogunbina, Adam Schwalje, Steven Losorelli, Stephen McInturff, Peter Steyger^ (standing), Lina Reiss^, and Amanda Lauer*^. *, Person is not deaf or hard of hearing; ^, person is in the STEMM for Students with Hearing Loss to Engage in Auditory Research (STEMM-HEAR) faculty. Photo by Chin-Fu Liu*.

Cambridge, MA, my main accommodation was note taking, with occasional one-to-one discussions with professors or graduate students. After college, I have been communicating in both LSL and American Sign Language (ASL), the latter of which enabled me to use ASL interpreters at the University of Pennsylvania, Philadelphia. These accommodations enabled me to complete my PhD thesis under the supervision of James Saunders, focusing on hearing restoration in birds following acoustic trauma.

One of the things I have learned from attending conferences and meetings is that there are often accommodations for pre-planned events. However, a major factor for effective scientific collaboration is impromptu conversations with colleagues at conferences. It is impossible to predict when or where these conversations will occur, much less request ASL interpreters. Recent technological advances now include speech-to-text apps for smartphones that could be used for these impromptu scientific discussions, although initial experience shows that these apps may incorrectly translate technical terms.

I have observed my D/HH peers with cochlear implants succeeding because they are better able to participate in discus-

sions, gaining a bigger picture of their scientific interests. This is different from family discussions because my immediate family would always get me involved. Until my marriage to a deaf woman, I was the only deaf member of my immediate family. Also, my nephew Robby has to fight all the time for his parents’ attention when his own family is having a discussion even though he has bilateral cochlear implants. He simply does not like to be left out. Nonetheless, I hesitate on having a cochlear implant myself because I am at peace with my disability. Perseverance and tenacity are key to a successful academic career. My primary interest in biology combined with my hearing loss to cement a lifelong interest in hearing research. No matter what happens in the future, it is important that hearing research gains more than one perspective, especially that provided by diverse professionals with their own hearing loss.

J. Tilak Ratnanather

Born in Sri Lanka with profound bilateral hearing loss, I benefited from early diagnosis and intervention (both of which were unheard of in the 1960s but are now common practice worldwide) that led my parents to return to England. At two outstanding schools for the deaf (Woodford School, now closed, and Mary Hare School, Newbury, Berkshire, UK), I developed the skills in LSL that enabled me to matriculate in mathematics at University College London, UK. More recently, I have benefited from bimodal auditory inputs via a cochlear implant (CI) and a digital hearing aid in the contralateral ear.

In the late 1980s, I was completing my DPhil in mathematics at the University of Oxford, Oxford, UK. One afternoon, when nothing was going right, I stumbled on a mathematical biology seminar on the topic of cochlear fluid mechanics. An hour later, I knew what I wanted to do for the rest of my life. I first did postdoctoral work in London, which gave me an opportunity to visit Bell Labs in Murray Hill, NJ, in 1990. This enabled me to attend the Centennial Convention of the Alexander Graham Bell Association for the Deaf and Hard of Hearing (AG Bell) in Washington, DC. There I heard William Brownell from Johns Hopkins University (JHU), Baltimore, MD, discuss the discovery of cochlear outer hair cell electromotility (see article by Brownell [2017] in *Acoustics Today*). A brief conversation resulted in my moving to JHU the following year to work as a postdoctoral fellow with Brownell.

It was at this convention that I came across a statement in the Strategic Plan of the newly established NIDCD (1989, p. 247).

“The NIDCD should lead the NIH in efforts to recruit and train deaf investigators and clinicians and to assertively pursue the recruitment and research of individuals with communication disorders. Too often deafness and communication disorders have been grounds for employment discrimination. The NIDCD has a special responsibility to assure that these citizens are offered equal opportunity to be included in the national biomedical enterprise.”

This enabled me to realize that I could become a role model for young people with hearing loss. Meeting Henry and Peter cemented my calling. My research in the auditory system began with models of cochlear micromechanics and now focuses on modeling the primary and secondary auditory cortices. I also mentored students and peers with hearing loss in STEMM. In 2015, these efforts were recognized with my receiving a Presidential Award for Excellence in Science, Mathematics and Engineering Mentoring (PAESMEM) from President Obama.

Today, more young people who benefited from early diagnosis and intervention with hearing aids and/or cochlear implants are now entering college. Many want to study the auditory system to pay forward to society. The PAESMEM spurred me to establish, with the cooperation of AG Bell, STEMM for Students with Hearing Loss to Engage in Auditory Research (STEMM-HEAR; deafearsScientists.org) nationwide. In recent summers, students worked at the Oregon Health and Science University, Portland; Stanford University, Stanford, CA; the University of Minnesota, Minneapolis; the University of Southern California, Los Angeles; and JHU. STEMM-HEAR exploits the fact that hearing research is at the interface of the STEMM disciplines and is a perfect stepping stone to STEMM. STEMM-HEAR is now exploring at how off-the-shelf speech-to-text technologies such as Google Live Transcribe and Microsoft Translator can be used to widen access in STEMM (Ratnanather, 2017).

Peter S. Steyger

Matriculating into the University of Manchester, Manchester, UK, in 1981 was a moment of personal and academic liberation. Higher education settings had seemingly embraced diversity, however imperfectly, based on academic merit. Finally, I could ask academic questions without embarrassing teachers lacking definitive answers. Indeed, asking questions where answers are uncertain or conventional wisdom insufficient led to praise from professors and the confidence to explore further, particularly via microscopy in my case. Nonetheless, I remained a “solitaire,” the only deaf undergraduate

in the zoology class of 1984 and indeed of all undergraduates in biological sciences between 1981 and 1984.

One strategy deaf individuals using LSL use is to read voraciously (to compensate for missed verbal information), and I had subscribed to *New Scientist*. An issue in 1986 invited applications to investigate ototoxicity (the origin of my own hearing loss as an infant) using microscopy under the direction of Carole Hackney and David Furness at Keele University, Staffordshire, UK. That synergy of microscopy, ototoxicity, and personal experience was electrifying and continues to this day. This synergy also propels other researchers with hearing loss to answer important questions underlying hearing loss. These answers need to *make rational sense* and not just satisfy researchers with typical hearing who take auditory proficiency for granted. As our understanding of the mechanisms of hearing loss grows, the more we recognize the subtler ways hearing loss impacts each of us personally or those we hold dear as well as society in general. Accessibility and effective mentorship are vital for inclusion and growth during university and postdoctoral training.

I now experience *age-related hearing* because new hearing technologies are personally adopted and am currently bimodal, using a CI in one ear and connected via Bluetooth to a hearing aid in the other. Each technological advance enabled the acquisition of new auditory skills, such as sound directionality and greater recognition of additional environmental or speech cues, contrasting with peers with age-related hearing loss unable or unwilling to adopt advances in hearing technology. Each advance in accessibility, mentorship, and technology accelerates the career trajectories of aided individuals. With the acquisition of each new auditory skill, I marvel anew about how sound enlivens the human experience.

Brad N. Buran

My parents began using Cued Speech with me following my diagnosis of profound sensorineural hearing loss at 14 months of age. Cued Speech uses handshapes and hand placements to provide visual contrast between sounds that appear the same on the lips. Because Cued Speech provides phonemic visualization of speech, I learned English as a native speaker. Although I received bilateral cochlear implants as a young adult, I still rely on visual forms of communication to supplement the auditory input from the implants.

Interested in learning more about my deafness, I studied inner ear development in Doris Wu’s laboratory at the National Institute on Deafness and Other Communication Disorders (NIDCD), Bethesda, MD, as an intern during high school.

This was followed by an undergraduate research project on the inner ears of deep-sea fishes in Arthur Popper's laboratory at the University of Maryland, College Park. These early experiences cemented my interest in hearing research, driving me to pursue a PhD with Charles Liberman in the Harvard-MIT Program in Speech and Hearing Bioscience and Technology, Cambridge, MA.

Although my graduate classmates were interested in Cued Speech, most assumed they would not have time to learn. Realizing that Cued Speech is easy to learn, one classmate taught himself to cue over a weekend. Being a competitive group, my other classmates learned how to cue as well. I truly felt part of this group because I could seamlessly communicate with them.

Many of my peers in auditory science are interested in my experience as a deaf person. Their questions about deafness are savvier than those I encounter on the street. For example, the speech communication experts ask detailed questions about Cued Speech (e.g., how do you deal with coarticulation?). The auditory neuroscientists who dabble in music try to write a custom music score optimized for my hearing.

As a deaf scientist, communication with peers is a challenge. Scientists often have impromptu meetings with colleagues down the hall. If I cannot obtain an interpreter, I have to rely on lipreading and/or pen and paper. Fortunately, the Internet has significantly reduced these barriers. Most scientists have embraced email and Skype as key methods for communicating with each other. In my laboratory, we use Slack, a real-time messaging and chatroom app for most communications. Likewise, the availability of cloud-based resources for teaching has streamlined the programming in the neuroscience course I teach. Although I still use interpreters during classes, the availability of email and online chatrooms has allowed me to hold "virtual" office hours without having to track down an interpreter each time a student wants to meet with me. In addition to advances in technology, the advocacy of my senior D/HH colleagues has lowered barriers by increasing awareness of hearing loss in academia and ensuring that conferences are accessible to researchers with disabilities.

Take-Home Message

Researchers with hearing loss, regardless of etiology, bring many benefits to auditory sciences. Their training and vocabulary enable more accurate, real-world descriptions of auditory deficits, advancing knowledge in auditory sciences and stimulating research into mechanisms and implications

of auditory dysfunction. Their interactions with hearing researchers provide teachable moments in understanding the real-world effects of hearing loss. The ability to succeed in research requires resilience and perseverance. This is particularly true for individuals with disabilities who must overcome additional barriers. When provided with the resources they need and treated with the respect and empathy that all individuals deserve, they can make remarkable contributions to STEM, especially in the auditory sciences.

More importantly, these researchers are changing perceptions about how those with disabilities can integrate with mainstream society. However, this integration is not automatic. The maxim espoused by George Bernard Shaw (1903), "The reasonable man adapts himself to the world; the unreasonable one persists in trying to adapt the world to himself," remains pertinent. The ability to recognize new and emerging technological advancements and utilize creative strategies in adapting to one's own disability leads to a greater quality of life and more successful careers regardless of profession.

Last but not least, we would very much appreciate readers to encourage colleagues, staff, and trainees with hearing loss to join our expanding group (deafearscientists.org). Increased visibility and contributions by those with hearing loss can only enhance advances by the field as a whole!

References

-
- Adler, H. J., Anbuhl, K. L., Atcherson, S. R., Barlow, N., Brennan, M. A., Brigande, J. V., Buran, B. N., Fraenzer, J.-T., Gale, J. E., Gallun, F. J., Gluck, S. D., Goldsworthy, R. L., Heng, J., Hight, A. E., Huyck, J. J., Jacobson, B. D., Karasawa, T., Kovačić, D., Lim, S. R., Malone, A. K., Nolan, L. S., Pisano, D. V., Rao, V. R. M., Raphael, R. M., Ratnanather, J. T., Reiss, L. A. J., Ruffin, C. V., Schwalje, A. T., Sinan, M., Stahn, P., Steyger, P. S., Tang, S. J., Tejani, V. D., and Wong, V. (2017). Community network for deaf scientists. *Science* 356, 386-387. <https://doi.org/10.1126/science.aan2330>.
- Brownell, W. E. (2017). What is electromotility? – The history of its discovery and its relevance to acoustics. *Acoustics Today* 13(1), 20-27. Available at <https://acousticstoday.org/brownell-electromotility>.
- National Institute on Deafness and Other Communication Disorders (NIDCD). (1989). *A Report of the Task Force on the National Strategic Research Plan*. NIDCD, National Institutes of Health, Bethesda, MD.
- Phillips, K. W. (2014). How diversity works. *Scientific American* 311, 42-47. <https://doi.org/10.1038/scientificamerican1014-42>.
- Ratnanather, J. T. (2017). Accessible mathematics for people with hearing loss at colleges and universities. *Notices of the American Mathematical Society* 64, 1180-1183. <http://dx.doi.org/10.1090/noti1588>.
- Shaw, G. B. (1903). *Man and Superman*. Penguin Classics, London, UK.

Selected publications by Adler, Buran, Ratnanather, and Steyger that are not cited in the article. The purpose of these citations is to give an idea of the work of each author.

- Adler, H. J., Sanovich, E., Brittan-Powell, E. F., Yan, K., and Dooling, R. J. (2008). WDR1 presence in the songbird inner ear. *Hearing Research* 240,

102-111. <https://doi.org/10.1016/j.heares.2008.03.008>.

Buran, B. N., Sarro, E. C., Manno, F. A., Kang, R., Caras, M. L., and Sanes, D. H. (2014). A sensitive period for the impact of hearing loss on auditory perception. *The Journal of Neuroscience* 34, 2276-2284. <https://doi.org/10.1523/jneurosci.0647-13.2014>.

Buran, B. N., Strenzke, N., Neef, A., Gundelfinger, E. D., Moser, T., and Liberman, M. C. (2010). Onset coding is degraded in auditory nerve fibers from mutant mice lacking synaptic ribbons. *The Journal of Neuroscience* 30, 7587-7597. <https://doi.org/10.1523/jneurosci.0389-10.2010>.

Garinis, A. C., Cross, C. P., Srikanth, P., Carroll, K., Feeney, M. P., Keefe, D. H., Hunter, L. L., Putterman, D. B., Cohen, D. M., Gold, J. A., and Steyger, P. S. (2017). The cumulative effects of intravenous antibiotic treatments on hearing in patients with cystic fibrosis. *Journal of Cystic Fibrosis* 16, 401-409. <https://doi.org/10.1016/j.jcf.2017.01.006>.

Koo, J. W., Quintanilla-Dieck, L., Jiang, M., Liu, J., Urdang, Z. D., Allensworth, J. J., Cross, C. P., Li, H., and Steyger, P. S. (2015). Endotoxemia-mediated inflammation potentiates aminoglycoside-induced ototoxicity. *Science Translational Medicine* 7, 298ra118. <https://doi.org/10.1126/scitranslmed.aac5546>.

Manohar, S., Dahar, K., Adler, H. J., Dalian, D., and Salvi, R. (2016). Noise-induced hearing loss: Neuropathic pain via Ntrk1 signaling. *Molecular and Cellular Neuroscience* 75, 101-112. <https://doi.org/10.1016/j.mcn.2016.07.005>.

Ratnanather, J. T., Arguillère, S., Kутten, K. S., Hubka, P., Kral A., and Younes, L. (2019). *3D Normal Coordinate Systems for Cortical Areas*. Preprint available at <https://arxiv.org/abs/1806.11169>.

Find us on Social Media!



ASA

Facebook: @acousticsorg

Twitter: @acousticsorg

LinkedIn: The Acoustical Society of America

Youtube: acousticstoday.org/youtube

Vimeo: AcousticalSociety

Instagram: AcousticalSocietyofAmerica

The Journal of the Acoustical Society of America

Facebook: @JournaloftheAcousticalSocietyofAmerica

Twitter: @ASA_JASA

Proceedings of Meetings on Acoustics

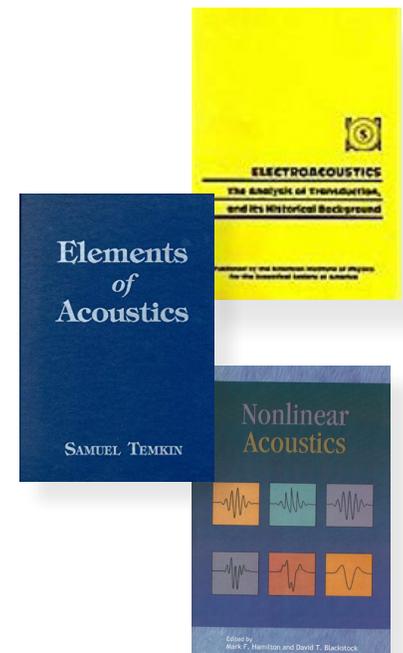
Facebook: @ASAPOMA

Twitter: @ASA_POMA

ASA Books available through Amazon.com

The **ASA Press** offers a select group of Acoustical Society of America titles at low member prices on Amazon.com with shipping costs as low as \$3.99 per book. Amazon Prime members can receive two-day delivery and free shipping.

For more information and updates about ASA books on Amazon, please contact the ASA Publications Office at **508-534-8645**.



ASA Press - Publications Office
P.O. Box 809
Mashpee, MA 02649
508-534-8645

International Student Challenge Problem in Acoustic Signal Processing 2019

Brian G. Ferguson

Address:

Defence Science and Technology (DST)
Group – Sydney
Department of Defence
Locked Bag 7005
Liverpool, New South Wales 1871
Australia

Email:

Brian.Ferguson@dsto.defence.gov.au

R. Lee Culver

Address:

Applied Research Laboratory
Pennsylvania State University
University Park, Pennsylvania 16802
USA

Email:

rlc5@psu.edu

Kay L. Gemba

Address:

Marine Physical Laboratory
Scripps Institution of Oceanography
University of California, San Diego
La Jolla, California 92093-0238
USA

Email:

gemba@ucsd.edu

The Acoustical Society of America (ASA) Technical Committee on Signal Processing in Acoustics develops initiatives to enhance interest and promote activity in acoustic signal processing. One of these initiatives is to pose international student challenge problems in acoustic signal processing (Ferguson and Culver, 2014). The International Student Challenge Problem for 2019 involves processing real acoustic sensor data to extract information about a source from the sound that it radiates. Students are given the opportunity to test rigorously a model that describes the transmission of sound across the air-sea interface.

It is almost 50 years since Bob Urick's seminal paper was published in *The Journal of the Acoustical Society of America* on the noise signature of an aircraft in level flight over a hydrophone in the sea. Urick (1972) predicted the possible existence of up to four separate contributions to the underwater sound field created by the presence of an airborne acoustic source. **Figure 1** depicts each of these contributions: direct refraction, one or more bottom reflections, the evanescent wave (alternatively termed the *lateral wave* or *inhomogeneous wave*), and sound scattered from a rough sea surface. Urick indicated that the relative importance of each contribution depends on the horizontal distance of the source from the hydrophone, the water depth, the depth of the hydrophone in relation to the wavelength of the noise radiated by the source, and the roughness of the sea surface.

The Student Challenge Problem in Acoustic Signal Processing 2019 considers the direct refraction path only. Other researchers have observed contributions of the acoustic noise radiated by an aircraft to the underwater sound field from one or more bottom reflections (Ferguson and Speechley, 1989) and from the evanescent wave (Dall'Osto and Dahl, 2015). When the aircraft flies overhead, its radiated acoustic noise is received directly by an underwater acoustic sensor (after transmission across the air-sea interface). When the aircraft is directly above the sensor, the acoustic energy from the airborne source propagates to the subsurface sensor via the vertical ray path for which the angle of incidence (measured from the normal to the air-sea interface) is zero. In this case, the vertical ray does not undergo refraction after transmission through the air-sea interface. The transmitted ray is refracted, however, when the angle of incidence is not zero. Snell's Law indicates that as the angle of incidence is increased from zero, the angle of refraction for the transmitted ray will increase more rapidly (due to the large disparity between the speed of sound travel in air and water) until the refracted ray coincides with the sea surface, which occurs when the critical angle of incidence is reached. The ratio of the speed of sound in air to that in water is 0.22, indicating that the critical angle of incidence is 13°. The transmission of aircraft noise across the air-sea interface occurs only when the angle of incidence is less than the critical angle; for angles of incidence exceeding the critical angle, the aircraft noise is reflected from the sea surface, with no energy propagating below the air-sea interface. The area just below the sea surface that is ensonified by the aircraft corresponds to the base of a cone; this area can be thought of as representing the acoustic footprint

Student Challenge Problem

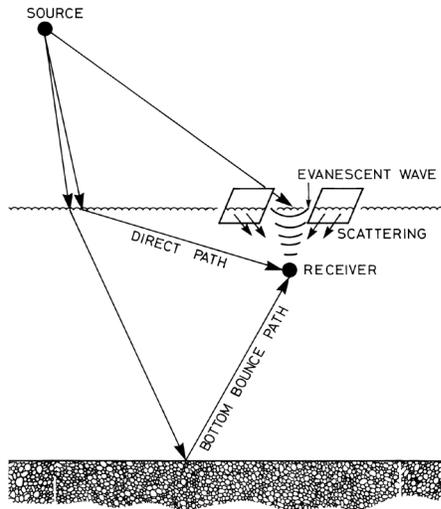


Figure 1. Contributions to the underwater sound field from an airborne source. After Urick (1972).

of the aircraft. The base of the cone subtends an apex angle, which is twice the critical angle, and the height of the cone corresponds to the altitude of the aircraft.

The first activity of the Student Challenge Problem is to test the validity of Urick's model for the propagation of a tone (constant-frequency signal emitted by the rotating propeller of the aircraft) from one isospeed sound propagation medium (air) to another isospeed sound propagation (seawater), where it is received by a hydrophone. Rather than measuring the variation with time of the received acoustic intensity as the acoustic footprint sweeps past the sensor (as Urick did), it is the observed variation with time of the instantaneous frequency of the propeller blade rate of the aircraft that is used to test the model. This is a more rigorous test of the model. The frequency of the tone (68 Hz) corresponds to the propeller blade rate (or blade-passing frequency), which is equal to the product of the number of blades on the propeller (4) and the propeller shaft rotation rate (17 Hz). For a turboprop aircraft, the propeller blade rate (or source frequency) is constant, but for a stationary observer, the received frequency is higher (commonly referred to as the "up Doppler") when the aircraft is inbound and lower ("down Doppler") when it is outbound. It is only when the aircraft is directly over the receiver that the source (or rest) frequency is observed (allowing for the propagation delay). The Doppler effect for the transit of a turboprop aircraft over a hydrophone can be observed in the variation with time (in time steps of 0.024 s) of the instantaneous frequency measurements

of the received signal, which is recorded in the file Time vs. Frequency Observations. This file can be downloaded at acousticstoday.org/iscpasp2019. The first record at time -1.296 s and frequency 73.81 Hz indicates that the aircraft is inbound, and for the last record at time 1.176 s and frequency 63.19 Hz, it is outbound.

Task 1

Given that a turboprop aircraft is in level flight at a speed of 239 knots (123 m/s) and an altitude of 496 feet (151 m); that the depth of the hydrophone is 20 m below the (flat) sea surface; that the isospeed of sound propagation in air is 340 m/s; and that in seawater, it is 1,520 m/s, the students are invited to predict the variation with time of the instantaneous frequency using Urick's two isospeed sound propagation media approach and comment on its goodness of fit to the measurements in the file.

Task 2

Figure 2 is a surface plot showing the beamformed output of a line array of hydrophones as a function of frequency (0 to 100 Hz) and apparent bearing (0 to 180°). This plot shows the characteristic track of an aircraft flying directly over the array in a direction coinciding with the longitudinal axis of the array. The aircraft approaches from the forward end-fire direction (bearing 0° ; maximum positive Doppler shift in the blade rate), flies overhead (bearing 90° ; zero Doppler shift), and then recedes in the aft end-fire direction (180° ; maximum negative Doppler shift). For this case, the bearing corresponds to the elevation angle (ξ), which is shown in **Figure 3**, along with the depression angle (γ) of the incident ray in air. The (frequency, bearing) coordinates of 32 points along the aircraft track shown in **Figure 2** are recorded in the file Frequency vs. Bearing Observations, which can be downloaded at the above URL. Each coordinate pair defines an acoustic ray. Similar to the previous activity, for Task 2, the students are invited to predict the variation with the elevation angle of the instantaneous frequency of the source signal using Urick's two isospeed media approach and to comment on its goodness of fit to the actual data measurements. The aircraft speed is 125 m/s, the source frequency is 68.3 Hz, and the sound speed in sea water is 1,520 m/s.

Task 3

To replicate Urick's field experiment, a hydrophone is placed at a depth of 90 m in the ocean and its output is sampled

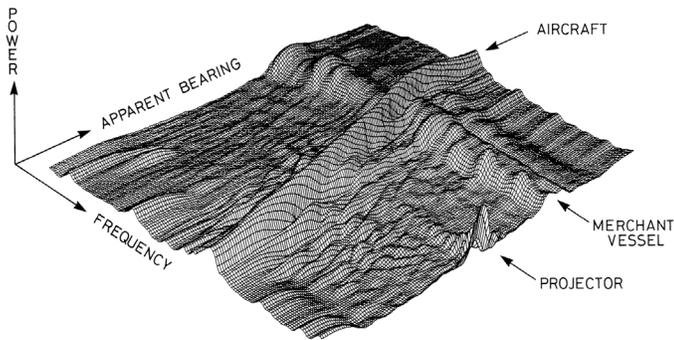


Figure 2. Variation with frequency and apparent bearing of the output power of a line array of hydrophones. Prominent sources of acoustic energy are labeled. After Ferguson and Speechley (1989).

at 44.1 kHz for 2 minutes, during which time a turboprop aircraft passes overhead. The sampled data are recorded in Waveform Audio File format (WAV) with the file name Hydrophone Output Time Series, which can be downloaded at the above URL. The students are invited to estimate the speed of the aircraft (in meters/second), the altitude of the aircraft (in meters), the source (or rest) frequency (in hertz), and the time (in seconds) at which the aircraft is at its closest point of approach to the hydrophone (i.e., when the source is directly above the sensor).

The deadline for student submissions is September 30, 2019, with the finalists and prize winners (monetary prizes: first place \$500; second \$300; third \$200) being announced at the 178th meeting of the Acoustical Society of America in San Diego, CA, from November 30 to December 4, 2019.

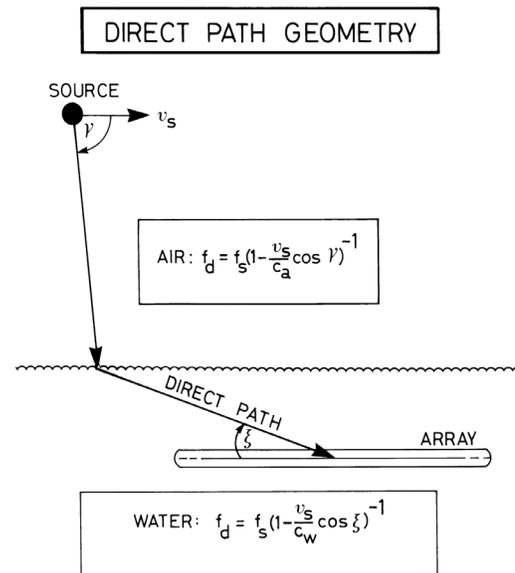


Figure 3. Direct refraction acoustic ray path and mathematical descriptions of the Doppler frequency (f_d), where f_s is the source frequency, v_s is the source speed, ξ is the elevation angle of the refracted ray, γ is the depression angle of the incident ray, and c_a and c_w are the speed of sound travel in air and water, respectively. The Doppler frequency and elevation angle are unique to each individual acoustic ray. After Ferguson and Speechley (1989).

References

- Dall'Osto, D. R., and Dahl, P. H. (2015). Using vector sensors to measure the complex acoustic intensity field. *The Journal of the Acoustical Society of America* 138, 1767.
- Ferguson, B. G., and Culver, R. L. (2014). International student challenge problem in acoustic signal processing. *Acoustics Today*, 10 (2), 26-29.
- Ferguson, B. G., and Speechley, G. C. (1989). Acoustic detection and localization of an ASW aircraft by a submarine. *The United States Navy Journal of Underwater Acoustics* 39, 25-41.
- Urick, R. J. (1972). Noise signature of an aircraft in level flight over a hydrophone in the sea. *The Journal of the Acoustical Society of America* 52, 993-999.

Planning Reading Material for Your Classroom?

Acoustics Today (acousticstoday.org) contains a wealth of excellent articles on a wide range of topics in acoustics.

All articles are open access so your students can read online or download at no cost. And point your students to AT when they're doing research papers!



Jozef J. Zwislöcki was born on March 19, 1922, in Lwow, Poland, and passed away on May 14, 2018, in Fayetteville, NY. He was a Distinguished Professor Emeritus of Neuroscience at Syracuse University, Syracuse, NY, a fellow of the Acoustical Society of America (ASA), and a member of the United States National

Academy of Sciences and the Polish Academy of Sciences. His large list of awards includes the first Békésy Medal from the ASA and the Award of Merit from the Association for Research in Otolaryngology. Zwislöcki's wide-ranging career focused on an integrative approach involving engineering, psychophysics, neurophysiology, education, and invention to advance our understanding of the auditory system and the brain. His early years were shaped by the events of World War II (his grandfather, Ignacy Mościcki, was the President of Poland from 1926 to 1939).

In 1948, Zwislöcki emerged on the scientific scene with his doctoral dissertation "Theory of Cochlear Mechanics: Qualitative and Quantitative Analysis" at the Federal Institute of Technology, in Zurich, Switzerland. The dissertation provided the first mathematical explanation for cochlear traveling waves. Recognition for this work led to positions at the University of Basel, Switzerland, and Harvard University, Cambridge, MA. In 1958, Zwislöcki moved to Syracuse University. There, in 1973, he founded the Institute for Sensory Research (ISR), a research center dedicated to the discovery and application of knowledge of the sensory systems and to the education of a new class of brain scientists who integrate the engineering and life sciences.

Throughout his career, Zwislöcki refined his theory of cochlear mechanics, modifying his model as new data became available and performing his own physiological experiments to test novel hypotheses. His contributions spanned the revolution in our understanding of cochlear mechanics, going from the passive broadly tuned cochlea observed by von Békésy in dead cochleas to the active sharply tuned response now known to be present in healthy cochleas, including the role of the tectorial membrane and outer hair cells in cochlear frequency selectivity. His psychophysical studies included

scaling of sensory magnitudes, both for the auditory system and other sensory systems; forward masking; just-noticeable differences in sound intensity; central masking; and temporal summation.

Zwislöcki searched for global interrelationships among psychophysical characteristics such as loudness, masking, and differential sensitivity, and their relationship to underlying neurophysiological mechanisms. He advanced our knowledge of middle ear dynamics, using modeling and measurements, and developing new instrumentation as required to improve our understanding of middle ear sound transmission and the effects of pathology. He performed studies of the stapedius muscle reflex both for its own sake and to analyze what this reflex implied about processing in the central nervous system. His work resulted in more than 200 peer-reviewed publications and numerous inventions, including the "Zwislöcki coupler." In his later years, he developed the Zwislöcki ear muffler (ZEM), a passive acoustic device that he anticipated would significantly reduce noise-induced hearing loss.

Zwislöcki loved skiing, sailing, trout fishing, horseback riding, and, of course, his wife of 25 years, Marie Zwislöcki, who survives him.

Selected Articles by Jozef J. Zwislöcki

- Zwislöcki, J. J. (1960). Theory of temporal auditory summation. *The Journal of the Acoustical Society of America* 32, 1046-1060.
- Zwislöcki, J. J. (2002). *Auditory Sound Transmission; An Autobiographical Perspective*. L. Erlbaum Associates, Mahwah, NJ.
- Zwislöcki, J. J. (2009) *Sensory Neuroscience: Four Laws of Psychophysics*. Springer US, New York, NY.
- Zwislöcki, J. J., and Goodman, D. A. (1980). Absolute scaling of sensory magnitudes: A validation. *Perception & Psychophysics* 28, 28-38.
- Zwislöcki, J. J., and Jordan, H. M. (1980). On the relations between intensity JNDs, loudness, and neural noise. *The Journal of the Acoustical Society of America* 79, 772-780.
- Zwislöcki, J. J., and Kletsy, E. J. (1979). Tectorial membrane: A possible effect on frequency analysis in the cochlea. *Science* 204, 638-639.

Written by:

Robert L. Smith

Email: RLSmith@syr.edu

Syracuse University, Syracuse, NY

Monita Chatterjee

Email: Monita.Chatterjee@boystown.org

Boys Town National Research Hospital, Omaha, NE



Did You Hear?

The Journal of the Acoustical Society of America

Special Issue on Ultrasound in Air

See these papers at:
acousticstoday.org/ultrasound-in-air

And, be sure to look for other special issues of *JASA* that are published every year.



Become a Member of the Acoustical Society of America

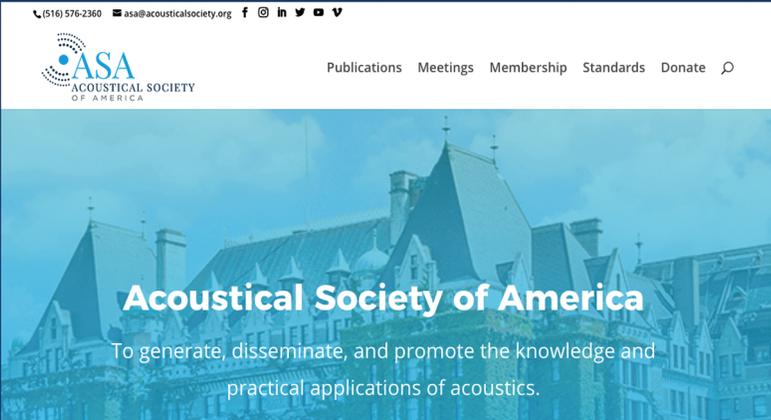
The Acoustical Society of America (ASA) invites individuals with a strong interest in any aspect of acoustics including (but not limited to) physical, engineering, oceanographic, biological, psychological, structural, and architectural, to apply for membership. This very broad diversity of interests, along with the opportunities provided for the exchange of knowledge and points of view, has become one of the Society's unique and strongest assets. From its beginning in 1929, ASA has sought to serve the widespread interests of its members and the acoustics community in all branches of acoustics, both theoretical and applied. ASA publishes the premier journal in the field and annually holds two exciting meetings that bring together colleagues from around the world.

Visit the **acousticalsociety.org** to learn more about the Society and membership.

NOW ACCEPTING VISITORS

acousticalsociety.org

Stop by our new website



(516) 576-2360 asa@acousticalsociety.org f in t o v

Publications Meetings Membership Standards Donate ↻

Acoustical Society of America

To generate, disseminate, and promote the knowledge and practical applications of acoustics.

Advertisers Index

Brüel & Kjaer Cover 4
www.bksv.com

Commercial Acoustics Cover 3
www.mfmca.com

Comsol Cover 2
www.comsol.com

G.R.A.S. Sound & Vibration..... page 3
www.gras.us

JLI Electronics page 76
www.jlielectronics.com

The Modal Shop page 76
www.modalshop.com

NGC Testing Services page 27
www.ngctestingservices.com

NTi Audio AG page 9
www.nti-audio.com

PAC International page 9
www.pac-intl.com

PCB Piezotronics, Inc. page 1
www.pcb.com

Scantek page 5
www.scantekinc.com

Business Directory

**MICROPHONE ASSEMBLIES
OEM PRODUCTION**

CAPSULES • HOUSINGS • MOUNTS
LEADS • CONNECTORS • WINDSCREENS
WATERPROOFING • CIRCUITRY
PCB ASSEMBLY • TESTING • PACKAGING

JLI ELECTRONICS, INC.
JLIELECTRONICS.COM • 215-256-3200

**INFRASOUND
MEASUREMENTS?**

Rent complete measurement solutions, including new microphones for ultra-low frequency, low noise, and more!

From Sensors to Complete Systems



SONIC BOOM



WIND TURBINE



TORNADO

hear what's new at
modalshop.com/low-freq
 513.351.9919

THE MODAL SHOP
MTS SYSTEMS CORPORATION

ACOUSTICS TODAY RECRUITING AND INFORMATION

Positions Available/Desired and Informational Advertisements may be placed in ACOUSTICS TODAY in two ways: — display advertisements and classified line advertisements.

Recruitment Display Advertisements:

Available in the same formats and rates as Product Display advertisements. In addition, recruitment display advertisers using 1/4 page or larger for their recruitment display ads may request that the text-only portion of their ads (submitted as an MS Word file) may be placed on ASA ONLINE — JOB OPENINGS for 2 months at no additional charge. All rates are commissionable to advertising agencies at 15%.

Classified Line Advertisements 2017 Rates: Positions Available/Desired and informational advertisements

One column ad, \$40.00 per line or fractions thereof (44 characters per line), \$200 minimum, maximum length 40 lines; two-column ad, \$52 per line or fractions thereof (88 characters per line), \$350 minimum, maximum length 60 lines. (**Positions desired ONLY** — ads from individual members of the Acoustical Society of America receive a 50% discount)

Submission Deadline:

1st of month preceding cover date.
Ad Submission: E-mail: dbott@aip.org; tel.: 516-576-2430. You will be invoiced upon publication of issue. Checks should be made payable to the Acoustical Society of America. Mail to: Acoustics Today, Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11747. If anonymity is requested, ASA will assign box numbers. Replies to box numbers will be forwarded twice a week. Acoustics Today reserves the right to accept or reject ads at our discretion. Cancellations cannot be honored after deadline date.

It is presumed that the following advertisers are in full compliance with applicable equal opportunity laws and, wish to receive applications from qualified persons regardless of race, age, national origin, religion, physical handicap, or sexual orientation.

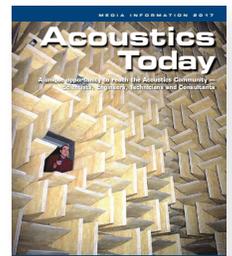
Advertise with Us!

Call Debbie Bott
 Advertising Sales Manager

516-576-2430

Advertising Sales & Production

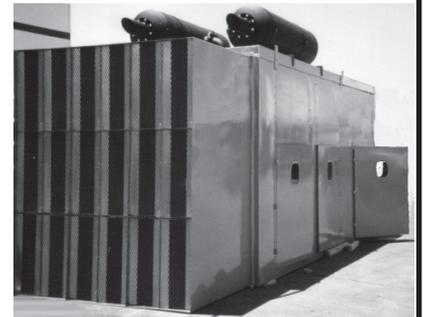
Debbie Bott, Advertising Sales Manager
 Acoustics Today, c/o AIPP Advertising Dept
 1305 Walt Whitman Rd, Suite 300
 Melville, NY 11747-4300
 Phone: (800) 247-2242 or (516) 576-2430
 Fax: (516) 576-2481 / Email: dbott@aip.org



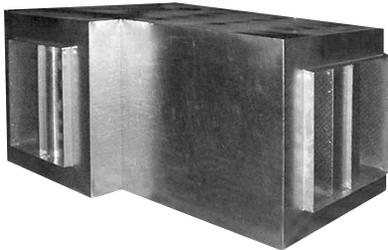
For information on rates and specifications, including display, business card and classified advertising, go to Acoustics Today Media Kit online at: https://advertising.aip.org/Rate_Card/AT2019.pdf or contact the Advertising staff.

PROVEN PERFORMANCE

For over 40 years Commercial Acoustics has been helping to solve noise sensitive projects by providing field proven solutions including Sound Barriers, Acoustical Enclosures, Sound Attenuators and Acoustical Louvers.



We manufacture to standard specifications and to specific customized request.



- Circular & Rectangular Silencers in Dissipative and Reactive Designs
- Clean-Built Silencers • Elbow Silencers and Mufflers • Independently Tested
- Custom Enclosures • Acoustical Panels • Barrier Wall Systems

Let us PERFORM for you on your next noise abatement project!



Commercial Acoustics

A DIVISION OF METAL FORM MFG., CO.

Satisfying Clients Worldwide for Over 40 Years.

5960 West Washington Street, Phoenix, AZ 85043

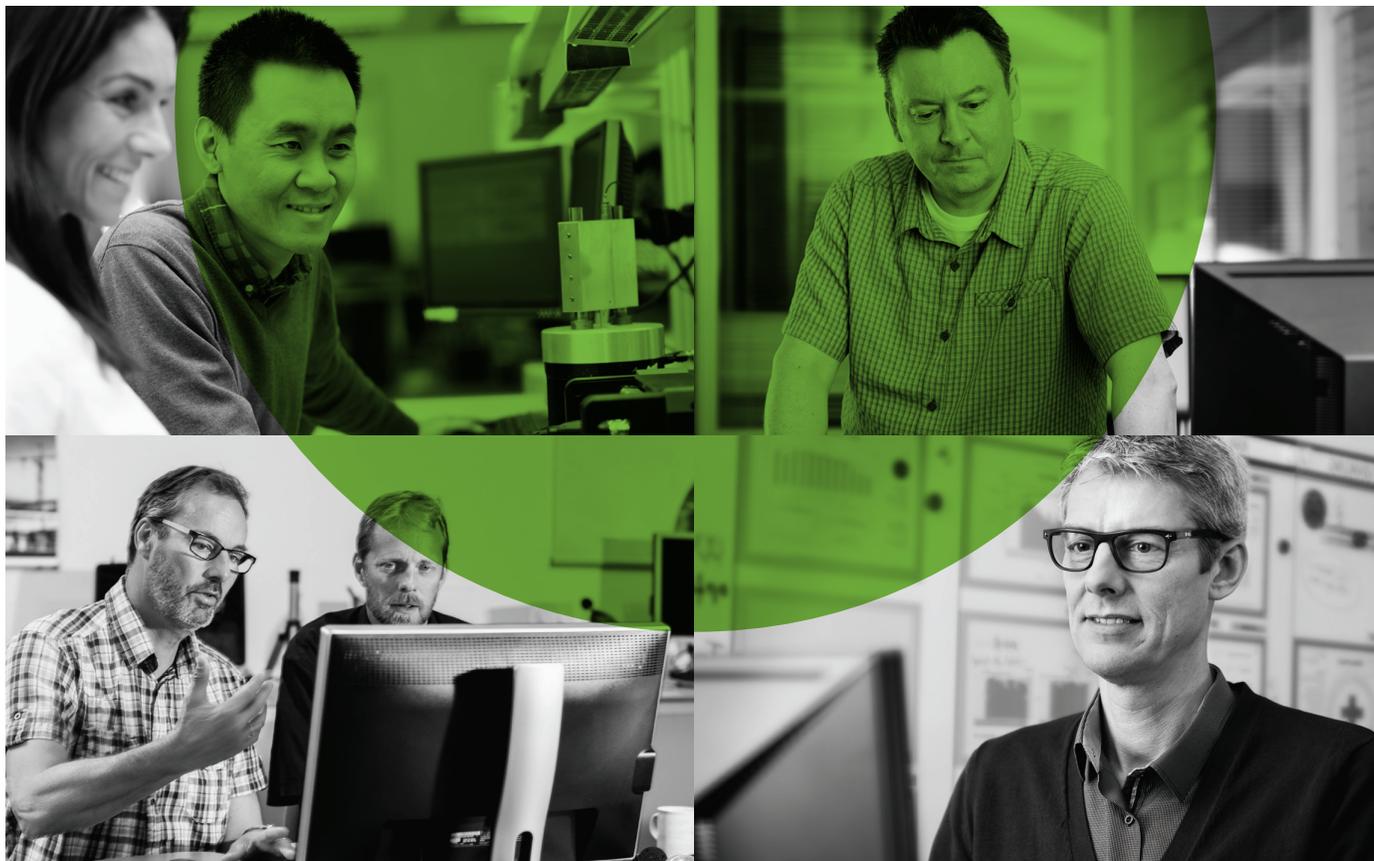
(602) 233-2322 • Fax: (602) 233-2033

www.mfmca.com

rbullock@mfmca.com

FLEXIBLE SOFTWARE PLATFORM

SOUND AND VIBRATION SOFTWARE THAT WORKS LIKE YOU WORK



BK CONNECT™ – A FLEXIBLE SOFTWARE PLATFORM DESIGNED AROUND YOUR NEEDS AND TASKS

Full of innovative features and functions, BK Connect – the new sound and vibration analysis platform from Brüel & Kjær – is designed around user workflows, tasks and needs, so you get **access to what you need, when you need it**. This user-friendly platform streamlines testing and analysis processes, which means you work smarter, with a high degree of flexibility and **greatly reduced risk of error**.



Brüel & Kjær 

BEYOND MEASURE

Brüel & Kjær Sound & Vibration North America, Inc.
3079 Premiere Parkway, Suite 120
Duluth, GA 30097
Telephone: 800 332 2040
bkinfo@bksv.com

bksv.com/bkconnect