Volume 14 | Issue 4 Winter 2018

Acoustics Today

ASA A Publication of the Acoustical Society of America

Archaeoacoustics: Re-Sounding Material Culture

Also In This Issue

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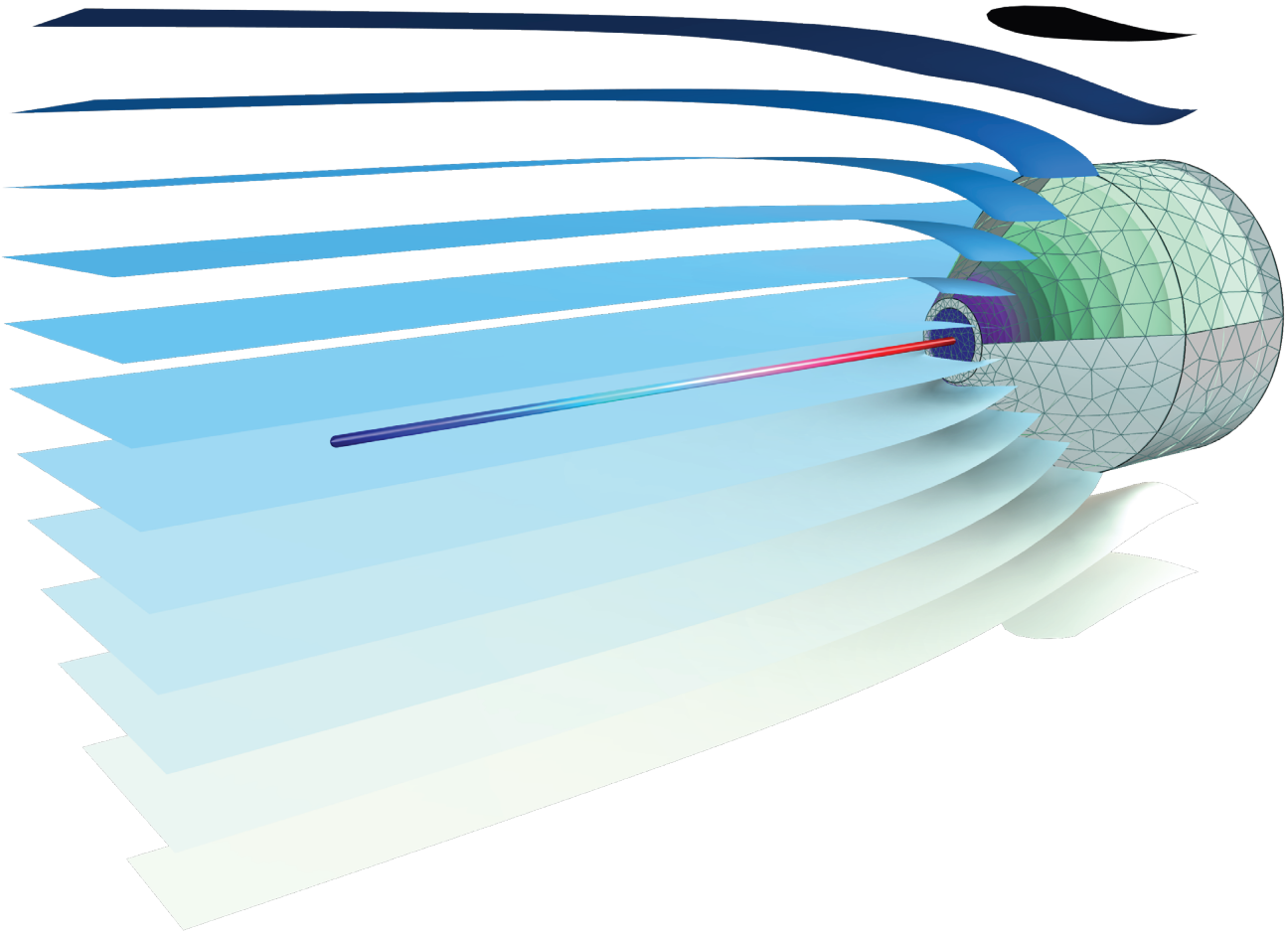

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Acoustics Today

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From "Archaeoacoustics: Re-Sounding Material Culture " by Miriam A. Kolar. Photography by José L. Cruzado Coronel, for Programa de Investigación Arqueológica y Conservación Chavín de Huántar, of Chavín pututu (Strombus Lobatus galeatus shell horn), currently on display in the Museo Nacional Chavín, Ancash, Perú.

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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/membership_and_benefits.**

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) Winter 2018, volume 14, issue 4, 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****.*

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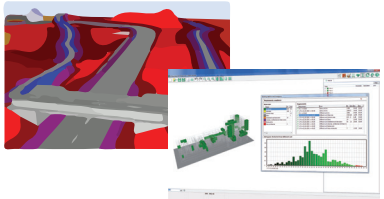

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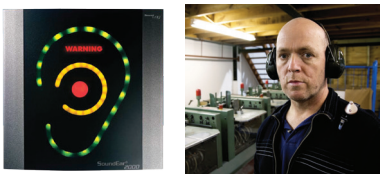

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Have you ever wondered how many people actually look at and read your scholarly writings? If you have, you likely realize that no matter how "great" one's paper is, readership is (with rare exceptions) quite small. Although it is hard to document "reads," if we use citations as a measure, the majority of papers only get cited a few times (and many not at all). Another measure is the number of downloads, and in many cases, such as *The Journal of the Acoustical Society of America* (*JASA*), one can easily find the number of times that a paper is viewed.

I think we would all agree that one's papers are seen by a very limited audience, and most often the audience has scholarly interests that are reasonably close to those of the authors. My point is to get readers of *Acoustics Today* (*AT*) to think about the readership of their scholarly works and to ask themselves whether they would like to write something that would be seen *and read* by a far broader and larger audience. In writing such an article, authors have the potential to reach an audience who would not normally see their work. This broad audience then becomes a way to "publicize" one's research area and bring new and exciting areas to the attention of a set of readers with very diverse interests. Put another way, this is an open invitation to members of the ASA to think about potentially writing an article for *AT*.

One of the best things about *AT* is that the audience is quite large. *AT* goes, *in hard copy*, to over 7,000 ASA members, and it is also read online by many people who are not ASA members, including students, journalists, and regulators. Moreover, articles are read by people from very diverse disciplines and not just the discipline of the authors. Indeed, I often hear from ASA members that they read *AT* from cover to cover and very much enjoy reading articles introducing them to research areas that they previously knew nothing about. (Indeed, one reason I enjoy editing *AT* is that I am learning so much about so many things that are new to me!) So, in writing for *AT*, you have an opportunity to reach a uniquely broad (and well-educated) audience.

In thinking about writing an article for *AT*, you need to keep several things in mind. First, and most important, articles have a broad focus. With *very rare* exception, articles do not focus on a single question or the work of one investigator or lab. Moreover, *AT* does not publish original research or preliminary results.

Second, an article for *AT*, although scholarly, is somewhat different from a scientific paper in that the audience is very broad. Thus, to be understood by our audience, authors must explain words and ideas that are likely to be totally "foreign" to a member of an ASA technical committee (TC) outside those of the authors. Although this kind of writing is a challenge, many authors have told me that this challenge ultimately turns out to be very enjoyable and intellectually exciting (and educational).

Third, once we agree on an article for a particular issue of *AT*, we must have the article at the agreed-to time. This is because we have only a few articles per issue, commit valuable space to them, and have no way of filling that space if we do not have the promised article.

I do hope many people reading this column will be intrigued with writing an article for *AT*. If so, please contact me. However, please, do not write the article before we "talk." This is because *AT* articles are by invitation only, and I cannot accept everything offered because I try to balance coverage in the magazine so that all TCs in the ASA are represented over time.

In this issue of *AT*, the first article, by Samira Anderson, Sandra Gordon-Salant, and Judy Dubno (former ASA president), discusses a topic of great relevance to those of us who are more "advanced" in years, and to others who will eventually get there: how hearing changes as we age. In the second article, Steve Greenberg writes about approaches to help people learn new languages. Steve shares insights into technology that run from the language labs I had in college (with reel-to-reel tape recorders) to very sophisticated apps for one's smart phone.

The third article illustrates one way I find articles. I am on the ASA Book Committee, and we had a proposal for a book on archaeoacoustics. I had never even heard the word and got very curious, so I wrote to Miriam Kolar and she agreed to do this article. Miriam uses some of her own work, and that of colleagues, to explain this fascinating field that merges classic archaeology and acoustics to better understand ancient civilizations.

Continued on page 9

Recent Actions on Financial Stewardship

In the 2015 Acoustical Society of America (ASA) Strategic Leadership Plan for the Future (see bit.ly/ASA-SLPF), one of the four primary goals specifically addresses financial stewardship: "ASA engages in wise, strategic stewardship to ensure sufficient resources to deliver maximum value both now and in the future." As with the other goals, a task force was formed to work with the ASA leadership to develop strategies toward safeguarding this goal. A number of changes impacting ASA finances have subsequently been implemented over the past few years, and I'd like to summarize those for you in this, my second president's column.

The current state of ASA finances is very strong, with a healthy amount of assets in reserve that continues to grow. A portion of those assets are held in restricted funds, overseen by the Acoustical Society Foundation Board. An overview of the assets, revenues, and expenses of the ASA was last provided by David Feit in the fall 2017 issue of *Acoustics Today* (see bit.ly/ASA-Finances-F17). David has been the ASA Treasurer since 2001, and the Society owes him a great deal of appreciation for his careful and steady guidance of ASA finances during these many years.

After lengthy discussions that began during the development of the Strategic Leadership for the Future Plan, it became clear that the ASA would be better equipped to meet anticipated financial challenges on the horizon if the Society added organizational capacity related to financial oversight. In 2015, only two administrative committees, the Audit Committee and the Investments Committee, dealt with ASA finances.

One of the first recommendations to be implemented out of the strategic plan was the formation of a Finance Committee that has a charge to make recommendations to the Executive Council on long-term financial decisions. The inaugural Finance Committee, chaired by former ASA President Anthony Atchley, began meeting regularly in 2016 and presented further recommendations to the Executive Council, including the following: (1) restructure the treasurer's position from one concerned heavily with administrative tasks to one focused more on policy and oversight; (2) hire an outsourced financial management firm to assist with handling the more administrative tasks; and (3) streamline financial reporting and forecasting to assist the Executive Council and other committees with being able to understand Society finances more efficiently.

The Executive Council has moved forward on all of these recommendations. At the 175th ASA meeting in Minneapolis, MN, in spring 2018, a vote was taken of the membership on a motion to change the Society's Bylaws. One of the most significant changes was to restructure the role of the ASA treasurer from an appointed position to an elected position with a three-year term of office and the option for one additional three-year term. That motion passed with a clear majority, and so for the first time in spring 2019, the ASA membership will be asked to vote for an elected treasurer every three years.

The new elected treasurer will play a major role in strategic oversight of ASA finances, beginning immediately after the spring 2019 meeting in Louisville, KY. In preparation for that, the bookkeeping and other accounting details are being transitioned to a financial management company, Kiwi Partners. I want to thank ASA headquarters and Jon Bara from Kiwi Partners for the extensive time they have put into making this a smooth transition. Much of that work has also been focused on improving the presentation and forecasting of financial data so that the volunteer leadership of ASA can more efficiently understand the state of ASA finances at a high level.

From a quick review of ASA finances and as David Feit pointed out in the article mentioned above, *The Journal of the Acoustical Society of America* (*JASA*) has been, and continues to be, the major source of revenue for the Society. The ASA Executive Council has for many years been vigilant on trying to understand and assess how recent and potentially rapid changes in the world of scholarly publishing may impact *JASA* and subsequently our Society's financial situation. ASA Editor in Chief James Lynch and the ASA Publications Office have been carefully monitoring the situation and taking steps to highlight and improve the strength of our journal (see bit.ly/AT-PubsQuality-S18). One recent action is that the ASA has entered into a five-year publishing partnership with the American Institute of Physics Publishing (AIPP) beginning in 2019. The terms of the partnership have been

thoroughly vetted by ASA headquarters, the Publications Office, the Finance Committee and other leadership. The partnership provides stability and greater incentives to both ASA and AIPP as we work collaboratively on marketing and maintaining the prominence of ASA publications. Strengthening ties between the ASA and AIPP will help both of our organizations in the years to come, as there remains much uncertainty regarding the future of scholarly publishing.

Finally, a discussion of ASA finances would be remiss without mention of the Acoustical Society Foundation Fund (ASFF) and the important role it plays in carrying out the mission of the ASA (see bit.ly/AcsSocFoundationFund). The ASFF is overseen by the Acoustical Society Foundation Board, currently chaired by former ASA President James Miller (2018-present). That committee of volunteers is charged primarily with assisting the Executive Council in acquiring, maintaining, and ensuring the correct expenditure of endowment funds. Structural changes have been made over the past few years to ensure that appropriate funds to support awards in the Society are all budgeted within the Foundation Fund.

The Foundation Board has also been leading new fundraising initiatives in support of the strategic goals of the ASA. A major ongoing fundraising effort is currently underway to endow fellowships for developing early-career leaders in our Society, as discussed in my last president's column (see bit.ly/AT-PresColumn-F18). Please join me and other members of the Executive Council in donating to the Campaign for ASA Early Career Leadership (see bit.ly/ASA-CAECL). All individual and corporate donors will be recognized on the campaign's website and in a commemorative booklet, to be published at the end of the campaign. Helping to establish these fellowships for early-career leaders in acoustics is one way that we can all be good stewards for the future of our Society.

In summary, ASA finances are in great shape, but we have concerns for the future as publication models change. Accordingly, and in response to the Strategic Leadership for the Future plan, many changes have taken place to help ensure that the ASA continues to enjoy a solid financial base for many years to come. I am very enthusiastic about the current momentum and positive trajectory of the Society with respect to financial stewardship. Thank you to the many members who have served on Executive Council over the past four years for their honest assessments, frank discussions, and decisive and careful actions with regards to the Society's finances. There is still more to be done, of course, as future ASA leadership teams explore options for diversifying revenue streams and articulate more clearly the Society's financial principles, related, for example, to risk management or underwriting the Society's activities. The ideas of members on these items are welcome, particularly as the ASA is about to enter the next cycle of strategic planning for the coming three to five years.

Finally, I would like to take this opportunity to remind you that I have tried to post regularly on assorted ASA topics to the online "ASA President's Blog" (see acousticalsociety.org/asa-presidents-blog) in an effort to keep the ASA membership informed and engaged. If there are questions or areas of particularly interest to you that I haven't addressed there or in these *Acoustics Today* columns, please do let me know (president@acousticalsociety.org). I also encourage all of you to become more involved with the Society by volunteering on any of our 41 technical and administrative committees. Please visit acousticalsociety.org/volunteer where you can learn more about these committees and submit a form to express your interest in volunteering.

From the Editor

Continued from page 6

The fourth article, by Caroline Lubert, took me back to my youth when Sputnik changed the world. I've never thought much about the sounds produced during a rocket launch and their implications, but I now realize that this is a major issue of concern in rocket science and engineering.

The final article came about when I saw an abstract for an ASA special session about thermophones, another word I did not know. I attended and then invited Nathanael Mayo to share this fascinating topic with the *AT* audience.

This issue also has a number of interesting "Sound Perspectives" essays. "Ask an Acoustician" focuses on Sandra Gordon-Salant (only by coincidence does Sandy have two articles in this issue). If you have ever wondered about the function and history of ASA chapters, be sure and read the essay by Kenneth Wood, Jr., and Jacob Mauck.

We also have an essay by D. Keith Wilson that should interest anyone who uses computational acoustics. Keith describes a new Technical Specialty Group in this area and provides ample reason why this area has become so important to all of us in the ASA.

Another thing I learned from doing this issue is that the ASA has a history of participation in the world-famous Intel Science and Engineering Fair. Through this, the ASA helps support upcoming acousticians. Our participation is wonderfully described by Jeffrey Vipperman. The ASA also supports science education through programs at ASA meetings for local high-school students. This is described in an essay by Tracianne Neilsen and L. Keeta Jones. Indeed, hopefully many ASA members will read both of these essays and reach out to the authors for ways they can get involved and help the ASA in these very important outreach efforts.

I close with a reminder that if you would like to discuss doing an article for *AT*, send me an email (apopper@umd.edu) with a mention of your idea and I will be in touch. I am particularly interested in articles on topics not yet covered in *AT* (all the articles in the past issues are available at acousticstoday.org) and articles from younger members.

Hearing and Aging Effects on Speech Understanding: Challenges and Solutions

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Development of effective, evidence-based solutions to overcoming communication barriers imposed by hearing loss is critical in our rapidly aging population.

Why Does Grandma Seem So Withdrawn Lately?

The scene is your annual Thanksgiving dinner. Your grandmother has been smiling throughout the dinner, but you can tell that she is not following the conversation. She often interjects the conversation with an off-topic comment, and when asked a question, she may respond with an answer that does not relate to the conversation. When someone asks, "Have you heard from Faith recently?" she might respond irritably, "Yes, I washed my face this morning." She is 85 years old, and you are concerned she may be losing cognitive function.

But is it her cognitive status, her hearing ability, or a combination of the two that prevents her from fully engaging in the conversation? The answers to these questions can be difficult to sort out. A hearing loss of just a mild-to-moderate degree can have a significant impact on one's ability to understand speech in background noise, even if communication in quiet, one-to-one settings remains unimpaired (Dubno et al., 1984).

However, cognitive processes such as working memory or speed of processing may also interfere with communication in background noise (Pichora-Fuller, 2003). Previous studies have shown that hearing loss is associated with cognitive decline (Lin et al., 2013). Clearly, this link between cognitive decline and hearing loss supports the importance of older adults, such as your grandmother, receiving a comprehensive audiological evaluation and suggestions for managing a hearing loss, if identified. Yet, older adults are often reluctant to pursue help for their hearing difficulties because of assumptions regarding the high cost of hearing aids or how the use of hearing aids may appear to others. Their friends may have shared negative experiences regarding hearing aid discomfort or inadequate performance in background noise. And, when an individual finally makes a decision to seek help, he or she may find that the communication barriers resulting from hearing loss can be difficult to overcome, even with appropriate diagnosis and management, for the reasons described in this article.

The Audiological Evaluation

What can the audiological evaluation reveal about your grandmother's ability to participate in a conversation at a crowded dinner table? The typical evaluation assesses peripheral hearing function in each ear by measuring detection of pure tones at a wide range of frequencies (0.25-8 kHz) and plotting these thresholds as an "audiogram" and by measuring the ability to understand one-syllable words presented in quiet at conversational levels ("speech recognition"). **Figure 1** displays pure-tone thresholds at a range of frequencies for a typical younger adult

with normal hearing and a typical older adult with hearing loss. The *y*-axis plots the levels (in dB hearing level [HL]) at which the listener can just barely hear the sound. In this case, the reference for decibels is the average thresholds for a large group of normal-hearing individuals who have no ear diseases, so smaller values (near 0 dB HL) indicate normal hearing and larger values indicate hearing loss. Note that the *y*-axis is reversed from the typical plotting convention so that the audiogram plots better thresholds nearer the top and poorer thresholds nearer the bottom.

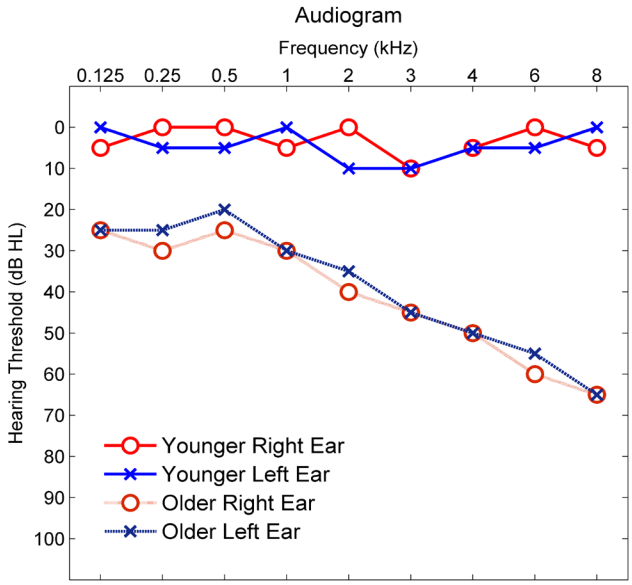

Figure 1. *Pure-tone thresholds as a function of frequency ("audiogram") for a typical younger adult with normal hearing (**solid lines**) and typical older adult with hearing loss (**dotted lines**). When thresholds are plotted on an audiogram, lower thresholds (near 0 dB hearing level [HL]) are near the top, indicating better hearing, and higher thresholds are near the bottom, indicating hearing loss (see text). According to the audiogram, the older adult has a mild hearing loss in the low frequencies, sloping to a moderately severe hearing loss in the high frequencies, which represents a typical pattern of age-related hearing loss.*

From the audiogram and one or more measures of speech recognition, the audiologist interprets the results to assess the type of hearing loss and the integrity of each part of the peripheral auditory system. The ear is composed of three main parts that contribute to audition: the outer ear (pinna and ear canal), the middle ear (eardrum, air-filled middle ear cavity, and middle ear bones), and the inner ear (cochlea and auditory nerve).

One issue might be a conductive hearing loss that usually indicates a pathology in the outer or middle ear, such as middle ear fluid or fusing of the middle ear bones, which prevents their movement in response to sound. These are examples of pathologies that prevent the conduction of sound through the middle ear. In contrast, a sensorineural hearing loss suggests a pathological condition in either the cochlea or the auditory nerve. The "sensory" component refers to the cochlea, and the "neural" component refers to problems primarily in the auditory nerve.

The sensory part of the inner ear is the cochlea that contains the organ of Corti. Lying on the organ of Corti are two types of sensory hair cells, the outer and inner hair cells. The inner hair cells transduce the sound and convert the sound vibrations into electrical energy. The outer hair cells serve as cochlear amplifiers to control the function of the inner hair cells. The gain of this amplifier can be increased or decreased by efferent neural connections that bring control of signals from the brain via olivocochlear reflexes that project from

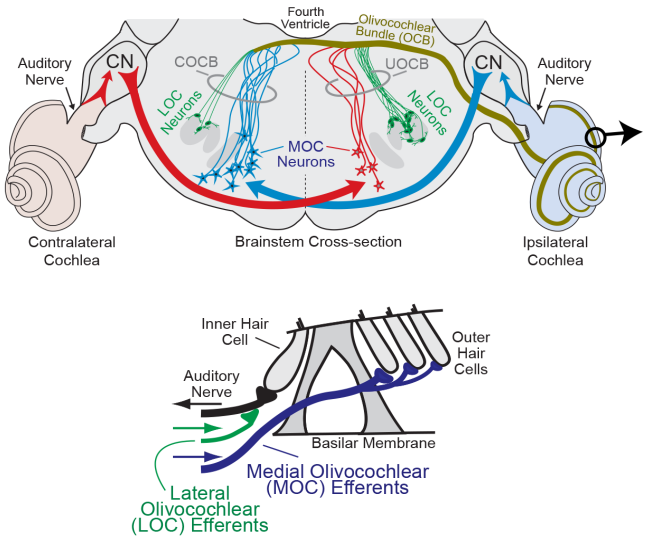

Figure 2. *A schematized anatomic view of the olivocochlear reflexes to the right cochlea. **Top:** outline of a transverse section of a cat brainstem showing the locations of lateral olivocochlear (LOC; **green**) and medial olivocochlear (MOC; **blue and red**) neurons. The pathways for the ipsilateral (stimulated cochlea; **blue**) and contralateral (unstimulated cochlea; **red**) MOC reflexes to the right ear join the olivocochlear bundle (OCB; **gold**). The axons from LOC and MOC neurons form the OCB, which is composed of crossed (COCB) and uncrossed (UOCB) components. The COCB is accessible near the fourth ventricle. CN, cochlear nucleus. The S-shaped gray structure is the lateral superior olivary nucleus, and the gray structure medial to it is the medial superior olivary nucleus. **Bottom:** schematic of the organ of Corti showing the main terminations of MOC neurons (**blue**) on outer hair cells and of LOC neurons (**green**) on the dendrites of auditory nerve fibers. From Guinan (2006), with permission.*

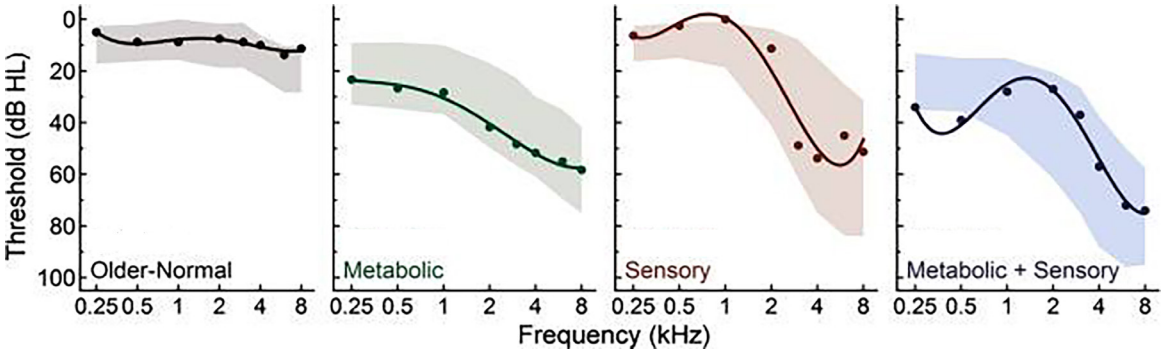

Figure 3. *Example audiograms are provided for the four major audiometric categories (phenotypes).* ***Shaded areas*** *correspond to the distribution of data labeled in these categories by experts. The "Older-Normal" and "Metabolic" phenotypes are similar to the audiograms for younger and older adults in* ***Figure 1****, respectively. The "Sensory" phenotype shows normal hearing in the low frequencies, dropping to a moderately severe hearing loss in the high frequencies, and the "Metabolic + Sensory" audiogram shows a moderate hearing loss in the low frequencies, dropping to a severe hearing loss in the high frequencies. Modified from Vaden et al. (2017), with permission.*

the superior olivary complex in the brainstem to the outer hair cells (depicted in **Figure 2**). Refer to Lonsbury-Martin et al. (2017) and Brownell (2017) for more detailed descriptions of cochlear mechanics.

The most common type of hearing loss in older adults is sensorineural hearing loss or presbyacusis. The audiograms of individuals with presbyacusis can be classified into four main categories or phenotypes (displayed in **Figure 3**): older normal, metabolic, sensory, and metabolic + sensory (Dubno et al., 2013). The metabolic category results from a loss of the endocochlear potential (positive voltage found in cochlear fluid). Because the endocochlear potential supplies power to the outer hair cells (see **Figure 2**), a decrease in the voltage can reduce the hair cells' ability to amplify incoming sounds. The metabolic category is typified by a flat mild hearing loss in the low frequencies gradually sloping to greater hearing loss in the high frequencies. The sensory phenotype is typified by a moderate-to-severe hearing loss in the high frequencies that may result from exposure to noise or drugs that are known to damage hearing. The metabolic + sensory phenotype is characterized by a low-frequency hearing loss similar to the metabolic phenotype and a moderate-to-severe high-frequency loss similar to the sensory phenotype. At present, the relationship between these hearing loss patterns and real-world hearing difficulty remains unknown, but in the future, the identification of a specific audiometric category may assist with determining possible hearing loss etiologies and with management planning.

The Audiogram Does Not Predict Speech Understanding in Noise

Patients with more severe hearing loss can expect greater speech understanding difficulties. A severe hearing loss would correspond to thresholds in the range of 60-90 dB HL. Nevertheless, the audiogram is not a good predictor of speech recognition in realistic conditions, such as in a noisy environment (Souza et al., 2007). Clinical measures of speech recognition in noise using words (Words-in-Noise [WIN] test; Wilson et al., 2007) or sentences (Quick Speech in Noise [QuickSIN] test; Killion et al., 2004) are becoming more common in audiological evaluations. But even these tests may not account for all of the difficulties experienced by older adults. Anderson et al. (2013a) investigated factors contributing to the variance in self-assessment of hearing ability in older adults using the Speech, Spatial, and Qualities of Hearing Questionnaire (SSQ; Gatehouse and Noble, 2004). In addition to the SSQ, the test measures included pure-tone thresholds, QuickSIN scores, and the frequency following response (FFR), a scalp-recorded measure of electrical activity that mirrors the timing and frequency aspects of the auditory stimulus. They found that the pure-tone thresholds and QuickSIN scores contributed to 15% of the variance in the SSQ score, and the FFR to a speech syllable accounted for an additional 15% of variance in the SSQ score. Therefore, factors in addition to peripheral hearing sensitivity, as indicated on the routine audiogram, may contribute to speech understanding difficulties.

Hidden Hearing Loss

Studies performed in the last decade have provided increasing evidence of peripheral hearing deficits that are not revealed in the audiogram or through otoacoustic emissions testing (a measurement of the sounds generated by the hair cells in the inner ear; Dubno et al., 2013). This type of deficit is now referred to as "hidden hearing loss" (Schaette and McAlpine, 2011). Aging may lead to one type of hidden hearing loss, a disruption of synapses (connections) between inner hair cells and auditory neurons that carry signals to the brain. This form of hidden hearing loss has been termed cochlear synaptopathy. Evidence of age-related cochlear synaptopathy was found in a mouse model (Sergeyenko et al., 2013). These older mice had normal-hearing thresholds, but neural firing to sounds above the threshold was reduced.

Varying degrees of cochlear synaptopathy may accompany sensorineural hearing loss, which may lead to frustrations experienced by patients and/or audiologists if audiometric thresholds do not predict success with management through hearing aids or cochlear implants. It is hoped that future research will be successful in developing proxy measures of synaptopathy that can be reliably obtained in a clinical setting.

"Don't Talk So Fast!"

The term "hidden hearing loss" may also be applied to processing deficits that affect the individual's ability to process the temporal or frequency properties of speech stimuli. For example, aging appears to have pronounced effects on the ability of the auditory system to preserve the precise timing characteristics of speech. We use timing cues, such as vowel duration, to distinguish words that differ in voicing, which occurs when the vocal folds of the larynx or voice box vibrate as air passes from the lungs to the oral cavity. For example, the vowel in "wheat" preceding the final voiceless consonant /t/ is shorter than the vowel in "weed" preceding the final voiced consonant /d/. In everyday conversational speech, the final consonant is not sufficiently audible for listeners to make that perceptual judgment without the vowel duration cue. Older adults have reduced ability to identify words on the basis of these and other temporal cues compared with younger adults (Gordon-Salant et al., 2008). In other words, an older adult would require a longer vowel duration to perceive "weed" versus "wheat." So, the next time you are speaking to your grandmother, try to slow down your rate of speech a bit to increase her ability to use these cues.

The perceptual consequences of disrupted temporal processing include a reduced ability to understand speech that is spoken rapidly or with an accent. As we age, we may find ourselves relying on open captions when watching many television shows or we may have difficulty understanding the younger relative who speaks rapidly. Decreased temporal processing may also affect the ability to understand speech in challenging listening environments, such as in background noise or in reverberant environments. These are the environments in which hearing aids are the least effective but where older adults report their greatest communication problems. As a result, older adults begin to avoid these troublesome listening situations and may avoid using their hearing aids.

"Why Are You Shouting?"

When the listener does not understand what was said, the natural tendency is for the speaker to repeat him/herself at a higher level. But the listener may then complain that the speaker is shouting. Individuals with hearing loss may need speech to be spoken at 70 dB sound pressure level (SPL; higher than average conversational speech) to understand the same message that might be understood at 30-40 dB SPL by someone with normal hearing; yet, at 100 dB SPL, speech becomes equally loud for individuals with either hearing loss or normal hearing. Thus, a person with normal hearing will have a dynamic range (difference between threshold and maximum tolerable loudness levels) of approximately 100 dB, but the individual with sensorineural hearing loss may have a dynamic range of 50 dB or less. This reduced dynamic range may lead to problems when trying to provide enough amplification to make soft sounds audible while limiting amplification for loud sounds so that they are not uncomfortably loud.

The loss of outer hair cells is one mechanism that may explain the reduced dynamic range observed in people with hearing loss. In the normal-hearing ear, the outer hair cells have abundant efferent connections that regulate the amount of amplification applied to sounds (see **Figure 2**). When outer hair cells are lost, low-level signals are not detected and there is no amplification provided to the signal by the outer hair cells. As the signal level is increased, there is a spread of excitation to neighboring hair cells, which then triggers cochlear amplification of the signal, resulting in an abrupt perceived increase in loudness. This rapid growth in loudness can occur when the sound level is increased by only 10 or 20 dB.

Another mechanism that may explain the reduced dynamic range is a disruption in the auditory system's maintenance of a stable firing rate over a period of time. The maintenance of a steady internal environment is known as homeostasis. A change in the balance of excitatory and inhibitory neurotransmission is one homeostatic mechanism that is associated with aging and hearing loss (Caspary et al., 2013). Communication between two neurons occurs through neurotransmission; neurons are more likely to fire when they receive excitatory input and less likely to fire when they receive inhibitory input. One possible result of the loss of inhibitory input with aging or hearing loss is an increase in spontaneous neural firing and exaggerated responses to auditory stimuli.

Electrophysiological (electroencephalographic [EEG]) studies have documented exaggerated responses to sounds presented at conversational listening levels of about 65-70 dB SPL. The FFR shows exaggerated subcortical responses to the speech envelope (slowly varying amplitude variations in speech) in older adults with sensorineural hearing loss (Anderson et al., 2013b). This exaggeration of responses to auditory stimuli may be especially pronounced in the cortex. Magnetoencephalographic (MEG) responses (observed on recordings of magnetic fields produced by electric currents in the brain) show overrepresentation of the speech envelope in older adults compared with younger adults (Presacco et al., 2016). Exaggerated responses to the speech envelope may help to explain why older adults find hearing aid-amplified sound so overwhelming when they first start wearing hearing aids.

"Why Is Speech So Unclear?"

Older adults often report they can hear the talker, but they cannot understand what is being said. Speech understanding may be reduced by deficits in the auditory system's ability to represent the timing and frequency cues of speech. The typical presbyacusic hearing loss compromises audibility in the high frequencies to a greater extent than in the low frequencies (see **Figure 1**). Therefore, merely amplifying the overall level of sound results in excessive amplification in the low frequencies where hearing is relatively normal and in perception of lower frequency background noise. Modern hearing aids are able to selectively amplify specific frequencies, within the limitations of the hearing aid microphone and circuitry. However, frequency selectivity (ability to detect differences in frequency) is often decreased in individuals with sensorineural hearing loss compared with individuals with normal hearing regardless of stimulus presentation level (Florentine et al., 1980). Therefore, the hearing aid user may not achieve maximum benefit from selective amplification of specific frequency channels.

The auditory system is organized tonotopically from the cochlea to the cortex; that is, low-to-high frequencies are represented in spatial order. For example, the cochlea is maximally responsive to high frequencies at the basal end (near the middle ear) and maximally responsive to low frequencies at the apical end (top of the cochlear spiral). This spatial organization is preserved throughout the auditory system. Hearing loss, however, may alter the tonotopic organization of central auditory structures.

For example, the C57 mouse model is used to study hearing loss effects because these mice commonly experience sensorineural hearing loss relatively early in the adult life span. C57 mice show disrupted tonotopic organization in the inferior colliculus, the auditory region of the midbrain, such that neurons that normally fire best to high-frequency sounds begin to respond more to low frequencies (Willott, 1991). Tonotopic changes may also occur in the auditory cortex of the brain. For example, when excessive noise damages hair cells in specific frequency regions in the cochlea (e.g., 3-6 kHz), stimulation with signals at these frequencies does not produce a response in cortical neurons in corresponding frequency regions but instead produces a response in neurons from adjacent cortical regions (Engineer et al., 2011).

Because of changes in frequency selectivity and tonotopicity, selective amplification of specific frequencies will not completely compensate for a decreased ability to discriminate between speech sounds based on subtle frequency differences. For example, the consonant /g/ has higher frequency energy than the consonant /d/. Although the two consonants differ in their place of articulation in the vocal tract, the place differences are not visible to the listener from viewing the talker's lips, and, therefore, the listener with hearing loss may have difficulty discriminating between words like "gust" and "dust" on the basis of frequency differences alone.

"Why Do I Still Have Trouble Understanding Speech with My Hearing Aids?"

Let us assume that your grandmother has been fit with hearing aids after being diagnosed with a mild-to-moderate hearing loss. You have been looking forward to the next family gathering, and you are hoping she participates more in the conversation. Your grandmother certainly seems more engaged, and yet she is still asking others to repeat what was

said, especially when the background noise levels are high, such as in a restaurant. It is important to remember that although hearing aid digital technology has improved dramatically in the last few decades, amplification cannot fully compensate for neural-processing deficits.

Hearing aid algorithms attempt to provide appropriate amplification to compensate for hearing loss at each frequency and to maintain sound levels within the dynamic range of the listener, based solely on the audiogram and measures of loudness discomfort. These algorithms also attempt to improve ease of listening in noise using directional microphone and noise reduction strategies so that the listener does not exert as much effort to understand what is being said. However, it can be difficult to evaluate the real-world effectiveness of amplification, particularly in a clinical setting. Audiologists use probe-microphone measurements to verify the appropriateness of the hearing aid fitting. During probe-microphone measurement, the audiologist places a thin tube in the ear canal a few centimeters from the ear drum. This tube is attached to a microphone, and the hearing aid is then placed alongside the tubing in the ear canal. Speech and other stimuli are presented at varying levels, and the audiologist determines if the amplified sound levels reaching the eardrum adequately compensate for hearing loss based on the pure-tone thresholds. Although verifying hearing aid fitting in this way is important, this measurement does not provide information about how speech is being processed by the inner ear, the central auditory system, and the brain.

Because of the limitations of probe-microphone measurements, interest in the use of EEG measures to assess the benefit of hearing aids is increasing. Several studies have focused on verifying detection of speech signals using EEG recordings in infants or other individuals who may not be able to provide feedback (e.g., Easwar et al., 2015). These measures may be useful in determining if amplification is providing sufficient audibility to detect speech consonants across a range of frequencies, but the effectiveness of EEG measures for providing information about the ability of the brain to discriminate between consonants has not yet been demonstrated (Billings et al., 2012). Furthermore, a detection measure may not be as relevant for an individual who can provide feedback about the audibility of different speech sounds.

What may be more useful is a measure that can provide information about the processing of conversational level speech rather than soft, threshold-level sounds. Age- and hearing (loss)-related deficits in temporal and frequency processing are observed at listening levels well above the speech threshold. A few studies have assessed the effects of amplification on the neural processing of conversational level speech stimuli (e.g., Jenkins et al., 2017), but more research is needed to determine if EEG measures can be used to assess improvements in neural processing in a clinical setting, with the goal of improved performance.

Figure 4. *Example of an in-ear electrocencephalographic (EEG) mount shown as a single earplug (**left**) and in the ear (**right**). Own work by Mikkelsen.kaare, Creative Commons Attribution-ShareAlike 4.0 International Public License (CC BY-SA 4.0). commons.wikimedia.org/w/index.php?curid=51268329.*

Despite advancements in digital noise reduction and directional microphone technologies, understanding speech in noise continues to be the greatest challenge experienced by individuals who use hearing aids. The main limitation is that current technology is unable to distinguish between a target talker, who should be amplified, and a background of multiple talkers, who should be attenuated. Generally, hearing aid algorithms use multiple microphones to focus amplification in the direction of whomever the listener is facing. It is reasonable to assume that the listener is usually facing the speaker. However, there may be times when the listener hears an interesting fragment from another speaker in the group and would prefer to listen in on that conversation without turning his/her head. Hearing aids generally will not be able to quickly and easily adjust to this scenario.

Future Directions

These limitations in current digital noise reduction technology have led to an interest in the development of cognitively driven or attention-driven hearing aids (Das et al., 2016). This research is based on evidence of the ability of EEG or MEG measurements to reveal the listener's object of attention (Ding and Simon, 2012; O'Sullivan et al., 2015). The idea behind the research is that discreet in-the-ear electrodes might be used to convey information to the hearing aid regarding the listener's focus of attention. The hearing aid processing algorithm would then selectively amplify the desired speech stream of interest to the listener. **Figure 4** shows an in-ear EEG mount. Another recent innovation is the "visually guided hearing aid prototype" that uses an eye tracker

mounted on a pair of eyeglasses to track the direction of the listener's gaze (Kidd et al., 2013), which may indicate the listener's focus of attention. The information regarding the direction of gaze is then used to maximize the directionality of a multimicrophone array.

The idea of "brain-controlled" hearing aids is certainly appealing. Nevertheless, the ability to benefit from this strategy is still limited by the ability of the auditory system to accurately process the amplified signal. As mentioned in **The Audiological Evaluation**, aging and hearing loss can disrupt the processing of the timing and frequency aspects of speech. The potential for neuroplastic changes in the aging auditory system has not yet been fully explored. There is some evidence that the use of hearing aids over time may improve the neural processing of speech signals, and that changes in cortical processing relate to improvements in cognitive function (Karawani et al., 2018), but more research is needed to explore the limits of neuroplasticity.

Outcomes may be improved if hearing aid use is supplemented with auditory training. But evidence for the potential benefits of auditory training to provide long-term improvement of perception and neural function has been mixed. A large-scale randomized control trial was conducted to evaluate the effects of supplementing hearing aid use with 10 hours of auditory training with Listening and Communication Enhancement Training (LACE) in 279 veterans and found that LACE training did not result in better outcomes than those obtained with standard-of-care hearing aid intervention alone (Saunders et al., 2016). Another study assessed the effects of 40 hours of auditory-based cognitive training in 29 older adults with and without hearing loss and found that the training improved performance on the QuickSIN and also reduced exaggeration of the speech envelope in older adults with hearing loss (Anderson et al., 2013b). Although the Anderson et al. study suggests that a sufficient number of hours of training may engender neuroplastic changes, the improvement in perception was relatively small and would not be considered clinically significant. It is possible that a behavioral measure of perception obtained in a laboratory setting does not capture training-related improvements that are experienced in real-life settings. Older adults with hearing loss expend more effort to understand speech, especially in noisy settings. An individual who expends considerable effort to understand what is said will not be able to maintain that level of effort over the long term. When effort cannot be sustained, speech perception may decrease and the individual begins to withdraw from the conversation. An objective measure of cognitive effort, such as pupillometry (a measure of pupil size and reactivity), may be a more sensitive assessment of training benefits than measures of speech recognition alone. Kuchinsky et al. (2016) found that twenty 90-minute sessions that trained word recognition in noise resulted in pupillometry changes that reflected a decrease in cognitive effort and improved word recognition in 29 older adults with hearing loss. Many questions remain unanswered regarding the potential for training to improve speech understanding in older adults. Studies are underway to assess the benefits of training that target age-related temporal processing deficits and auditory-cognitive interactions in older adults. A better understanding of training strategies that engender neuroplastic changes in older adults should lead to better outcomes and improved communication and social function in older adults.

Summary

Age-related hearing loss has many potential consequences for the quality of life, including social withdrawal and possible loss of cognitive function. It is therefore important to provide timely audiological assessment and management to individuals who appear to be having difficulty hearing and understanding speech. Nevertheless, aging brings additional challenges to identifying the source of speech-understanding problems, including disruptions in the transmission or processing of speech stimuli that can occur at all levels of the auditory system and the brain. Age-related cognitive decline may also contribute to speech-understanding problems. Therefore, it is imperative to identify and manage hearing loss to minimize the impact of cognitive decline. These disruptions may limit the benefit that can be obtained from hearing aid amplification and auditory training. Research is ongoing to optimize hearing aid technology using neural feedback regarding the listener's focus of attention. The development of effective auditory training programs may also improve hearing aid outcomes. Improved assessment and management protocols should improve the ability of older adults (including your grandmother) to maintain a healthy, active social life despite hearing loss.

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BioSketches

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Statement of Ownership, Management, and Circulation

Publication title:	*Acoustics Today*
Publication number:	1557-0215
Filing Date:	9/18/18
Issue frequency:	Quarterly
No. of issues published annually:	4
Annual Subscription price:	$5
Complete mailing address of known office of publication:	AIP Publishing LLC, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300
Contact person	Waylon Butler 516-576-2278

Full names and complete mailing addresses of publisher, editor, and managing editor:
Publisher: AIP Publishing LLC, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300
Editor: Dr. Arthur N. Popper, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300
Owner: Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300
Known bondholders, mortgagees, and other security holders owning or holding 1 percent or more of total amount of bonds, mortgages, or other securities: None

Tax Status: Purpose, function and nonprofit status of this organization and the exempt status for Federal income tax purposes: Has Not Changed During Preceding 12 Months
Publication title: *Acoustics Today*

Issue date for circulation data below: July 2018

Extent and nature of circulation:	Average No. of Copies Each Issue During Preceding 12 Months	No. of Copies of Single Issue Published Nearest to Filing Date
Total number of copies (Net press run)	7158	6960
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Paid, Mailed In: County		
Paid distribution Outside the Mails including Sales Through Dealers and Carriers, Street Vendors, Counter Sales, and Other Paid Distribution Outside USPS	2363	2168
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Total Distribution	7155	6963
Copies not Distributed	3	0
Total	7158	6963
Percent Paid	88.71%	88.47%

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18. Signature and Title of Editor, Publisher, Business Manager, or Owner: Roy Lavenson, CFO, AIP Publishing LLC — Date: 9/24/18

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Deep Language Learning

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How technology enhances language instruction.

Technology Is Transforming How Students Learn a Foreign Language

In *Star Trek*™ and other science fiction, alien civilizations communicate via a universal translator that seamlessly translates from one tongue to another, making alien-language instruction superfluous. Unfortunately, such flawless universal translation is unlikely to arrive anytime soon (despite recent advances).

So, at least for now, the most effective path for communicating in a foreign tongue is through instruction. Traditional language pedagogy emphasizes classroom and laboratory practice of vocabulary, grammar, and pronunciation. Lessons are highly structured, with students practicing language skills in class and laboratory. Feedback is offered mostly through exams and drills. However, this classical approach has serious drawbacks, especially when it focuses on declarative knowledge of grammar and vocabulary to the exclusion of conversational skills and comprehension.

Although the ambitious student might achieve conversational fluency by living in a foreign community, this option is unavailable to many. Fortunately, curricula are beginning to incorporate more naturalistic approaches to language learning, powered by technology. The long-term goal is to emulate real-world learning in ways that are effective, economical, and enjoyable.

For computer-assisted language learning (CALL), the "holy grail" is courseware that simulates what a student might experience living in a foreign land. In this virtual community, the student would converse in the target language and receive feedback on ways to improve. Although this pedagogical nirvana won't happen anytime soon, several advances bring it closer to reality. Among these are

(1) powerful, inexpensive computing residing in the "cloud," using a multitude (often thousands) of machines (usually graphical processing units [GPUs]) and abundant memory that mobile devices (e.g., smartphone, tablet) and computers can access easily;
(2) large amounts of online data to "train" pattern classifiers known as artificial neural networks (ANNs);
(3) cloud-based "deep learning" neural networks (DNNs). These are especially powerful ANNs that contain many (often dozens of) hidden layers and intricate connection topologies. A layer is "hidden" if it lies between the input stage of the ANN and its output (i.e., classifier outcome). Each hidden layer adds a level of processing that facilitates the "learning" (through adjustment of activation weights) of features critical for successful classification;
(4) DNN-trained automatic speech recognition (ASR) and speech synthesis (TTS) that deliver a quality and naturalness close to what humans achieve in many (though not all) languages. Many companies (e.g., Amazon, Apple, Google, and Microsoft) use the technology to interact with customers and clients. The data collected are used to further enhance the technology; and

(5) cloud-based virtual and augmented reality applications that extend or replace the user's physical environment through simulation of a variety of situations and environments.

These, along with advances yet to come, will transform the learning experience, not only for language instruction but also for pedagogy in general.

What is the current state of language learning technology, and where is it heading? Before answering, let's first review the history of CALL (Bax, 2003).

A Brief History of Computer-Assisted Language Learning

Computers were introduced into language instruction around 1960 to supplement programmed classroom instruction. Although the technology was primitive by today's standards, early CALL projects demonstrated a potential for enhancing the pedagogical experience. One example is the Programmed Logic for Automatic Teaching Operations (PLATO) Project (University of Illinois at Urbana-Champaign), which included online testing, tutoring, and chat rooms.

Over the years, the quality of CALL improved, driven by advances in interactive media and technology (Warschauer and Healy, 1998). In the 1960s and 1970s, CALL focused on drill and practice lessons in which a computer presented a stimulus and the student responded with (hopefully) the correct response. This was the "structural" (or "restricted") phase of CALL. Beginning in the late 1970s and extending through the early 1990s, CALL entered its "communicative" phase, which emphasized more natural ways of speaking and listening.

With the advent of the World Wide Web and multimedia technology in the 1990s, CALL entered its "integrative" phase, in which the pedagogy was incorporated into a broad range of communication scenarios representative of daily life. During this time, CALL applications offered graphics, animation, audio, and text, all in lessons that combined speaking, listening, reading, and writing (Chapelle and Sauro, 2017).

The key to effective language learning is for the student to use the foreign language as much as possible. Constant practice and feedback is essential. A shortage of language instructors and classroom time makes a compelling case for CALL because it offers instruction anytime, anywhere. Although CALL was originally designed for desktop and laptop computers, its future likely lies with smartphones, tablets, and other mobile devices (e.g., virtual reality [VR] goggles and artificial intelligence [AI]-enabled eyewear).

Computer-Assisted Pronunciation Evaluation and Training

Pronunciation training is where CALL has long deployed cutting-edge technology (Eskenazi, 2009). Several early projects used speech technology to evaluate a student's fluency, pronunciation proficiency, and comprehension. An example is SRI's Autograder project in which Japanese students were evaluated on their ability to speak English intelligibly. An algorithm was developed to emulate intelligibility judgments of native speakers but lacked remedial feedback. Some of this technology was incorporated into PhonePass™, an automatic system for evaluating a student's fluency and proficiency in English (Bernstein and Cheng, 2007).

Both academic (e.g., Carnegie-Mellon, Hong Kong, MIT, Nijmegen, KTH Stockholm) and commercial (e.g., Carnegie Speech, Duolingo, Rosetta Stone®, SRI, Transparent Language®) teams have developed technology that evaluates pronunciation using methods adopted from ASR. At first glance, ASR appears a perfect match for CALL. In place of a language teacher, why not leave the tedium of tutoring to an algorithm embedded in the cloud? It's available 24/7, never tires or sickens, and doesn't go on vacation. However, ASR-based CALL has its drawbacks. For one, ASR doesn't classify individual speech sounds with great precision (e.g., Greenberg and Chang, 2000). Like humans, automatic systems don't decode speech sound by sound but rather rely on clever engineering to infer what the speaker said (or should have said). They do so by culling information from a variety of nonacoustic sources (e.g., location, email, online searches) to supplement the acoustic signal. Although fortuitous for conventional ASR (e.g., Amazon's Alexa, Apple's Siri, and Google Voice), such supplementation can be a serious drawback for CALL applications. This is due to the uncertainty surrounding the identity of specific speech sounds (a.k.a. "phonetic segments" or "phones"). To better understand the problem, let's consider a hypothetical example. The word "pan" consists of three phonetic segments represented by the symbols [p], [æ], and [n] (brackets denote individual segments). An ASR system might correctly identify the initial and final consonants ([p] and [n]) but misidentify the vowel [æ], in which case the word initially "recognized" is "pin" rather than (the spoken word) "pan." However, the vowel's misclassification would probably be overridden by a

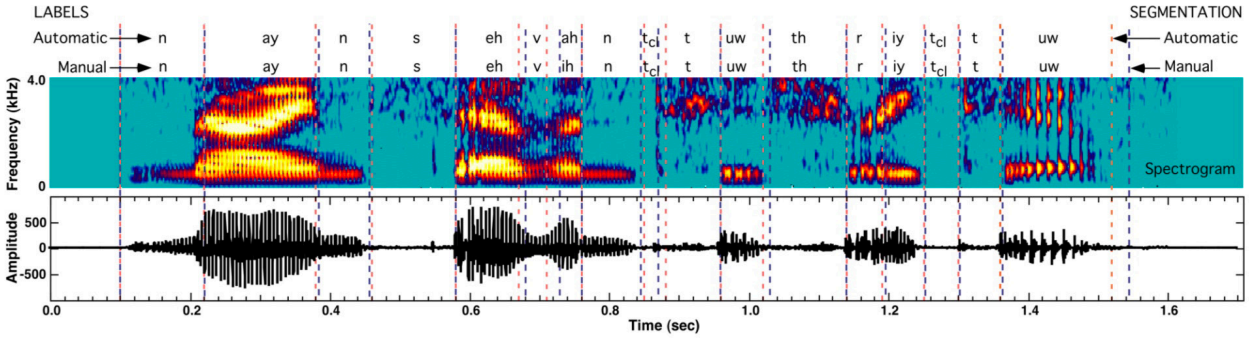

Figure 1. *Alignment of spoken material ("nine," "seven," "two," "three," "two") from the Oregon Graduate Institute "Numbers" corpus (Cole et al., 1994).* ***Top:*** *phonetic labels are similar to those used in the TIMIT corpus (Zue and Seneff, 1988);* ***middle:*** *spectrographic (time versus frequency) representation of the speech signal;* ***bottom:*** *speech pressure waveform. The "automatic" labeling and segment boundaries are analogous to an alignment. The "manual" labels and segment boundaries were provided by a trained phonetician. Reprinted from Chang et al. (2000), with permission.*

"language" model (Chelba and Jelinek, 2000) that factors in semantic context and lexical co-occurrence statistics to identify the word as "pan." ASR-based CALL requires other strategies to compensate for such phonetic imprecision, especially human listener judgments. However, such compensatory methods may themselves compromise the evaluation's accuracy.

An early example of ASR-based CALL was the Voice Interactive Language Training System (VILTS). A student's pronunciation was evaluated by comparing how well the ASR system performed relative to native speakers at a fine-grained level of analysis. The speech signal was partitioned into a sequence of basic sounds ("phones" or "phonetic segments"), and the results of a segment-based ASR system compared. Human raters graded a portion of the student material, and these data served as a baseline for normalizing the ASR-based scores. The system did not offer feedback on how to improve pronunciation (Neumeyer et al., 2000).

VILTS formed the foundation for another SRI system, EduSpeak® (Franco et al., 1999), which evaluates a student's pronunciation. The system comprises several stages: (1) segmentation and labeling (a.k.a. an "alignment") of individual phonetic segments (see **Figure 1**); (2) a measure of the distance between a student's speech and a native-speaker model (based largely on the similarity of their frequency spectra); (3) a comparison of automatically aligned phonetic-segment durations that takes the student's speaking rate into account; and (4) human listener evaluations for calibration. EduSpeak does not require a word transcript but is restricted to languages for which it has been explicitly trained.

Other groups (e.g., Witt and Young, 2000) have also deployed ASR for CALL. Many systems use human listener-based calibration to compensate for the imperfections of ASR. But, as Witt (2012) points out, even human listeners don't necessarily agree on the fine-grained quality of pronunciation (at the segment level), so why should machines be held to a higher standard?

Despite such caveats, several language programs, including Rosetta Stone and Carnegie Speech's NativeAccent®, do offer feedback at the word level (using ASR-based models) that students have found helpful. NativeAccent® also provides rudimentary diagrams of the vocal apparatus as part of its feedback.

Alternatives to ASR-based CALL compare a student's pronunciation to a native speaker's (or rather, a composite model based on a variety of speakers). The more similar the pronunciation of the two, the more intelligible the student's speech is likely to be.

Such a comparison involves both signal processing and acoustic analysis, and includes the following steps.

(1) Phonetic (and other forms of) feature extraction based on a range of spectral and temporal properties for classifying phonetic segments and/or linguistically relevant elements. The most frequent features are (a) a coarse snapshot (25 ms wide) of the acoustic frequency spectrum computed approximately every 10 ms (e.g., Mel Cepstral Frequency Coefficients [MFCCs]; Davis and Mermelstein, 1980); (b) a broadband frequency analysis with relatively fine temporal resolution (a spectrogram as in **Figure 1**); (c) temporal dynamics (velocity ["delta"] and acceleration ["double-delta"] features) of the spectrally filtered speech waveform (Furui, 1986), phonetic-segment and syllable duration as well as the trajectory of the fundamental frequency (pitch contour). A system may also "discover" the most relevant parameters through a process of "feature selection" (e.g., Li et al., 2018).

(2) "Alignment" of a student's speech. This produces a representation of the signal as a sequence of speech sound labels (e.g., [p], [ae], [n]) along with the start and end points of each sound in the speech waveform (these are approximate markers of where a speech sound is likely to begin and end).

(3) A word transcript is often required so that the aligner knows in advance the likely speech sounds and their sequence. The alignment (**Figure 1** shows an example) is based (in part) on acoustic models for each speech sound, often in the context of the sounds that precede and follow. In some systems, such as EduSpeak®, the student's speaking rate (in segments or syllables per second) is estimated as a way of improving the accuracy of the phonetic boundaries.

(4) Dynamic time warping (DTW; Sakoe and Chiba, 1978) is a method for aligning the student's speech, speech sound by speech sound, with a native-speaker composite model. DTW adjusts the segment boundaries to optimize the correspondence between the student and native-speaker model so that a "fair comparison" can be made at the phonetic-segment, syllable, and word levels (**Figure 2**).

(5) A distance metric that quantifies how similar the student's speech is to a native speaker (or rather a composite model comprising many native speakers). The features used for comparison are primarily spectral but may also incorporate dynamic and other temporal properties.

(6) The intrinsic variability of speech, particularly pronunciation, presents a major challenge for CALL technology. To simplify the comparison between the student's utterance and that of a native-speaker model, the analysis recasts the fine-grained spectral and temporal analyses into a form more amenable to quantification.

Because ASR systems have traditionally treated speech as a sequence of short-duration speech sounds (i.e., phonetic segments), it is this analytical framework that is most often used. However, word-level models are becoming increasingly popular and may replace segment models soon.

Three types of pronunciation errors account for most of the pronunciation problems students experience. These mostly occur at the level of individual speech segments (although some problems pertain to syllable prominence and duration). In the discussion that follows, a segment error is <u>underlined</u> to distinguish it from correctly articulated sounds.

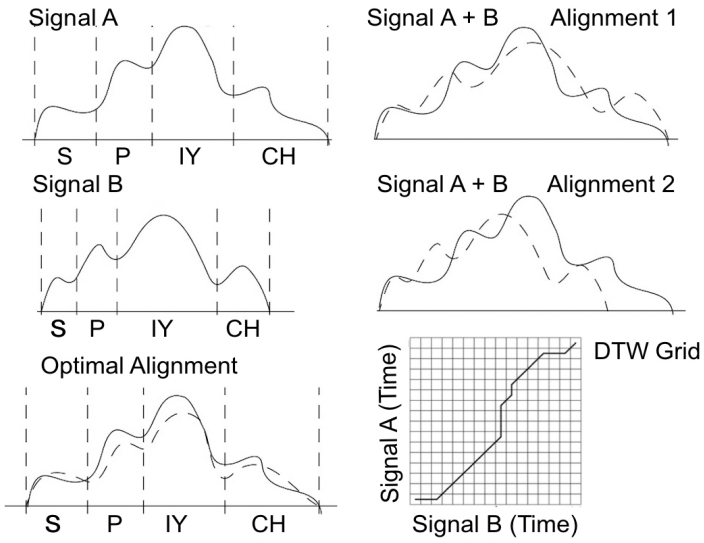

Figure 2. A highly simplified illustration of dynamic time warping (DTW) to achieve a "fair" comparison of two aligned speech signals. Signals A and B are two instances of the word "speech" spoken by two individuals. Dynamic time alignment iteratively warps the time axis of Signal B until it finds the closest match (in terms of time and spectrum) of the two signals.. The DTW grid shows a hypothetical time warping of Signal B (relative to Signal A) to achieve a quantitatively optimal alignment (i.e., the closest spectrotemporal match) of the two. Signal A and B alignment adapted from Zeng (2000) and DTW Grid adapted from Salvador and Chan (2007), with permission.

A "substitution" error would be one where the student pronounces the English word "land" as "<u>l</u>end." An "insertion" would occur if the student pronounces "land" as "land<u>s</u>." A "deletion" would occur if "land" were pronounced as "lan" (where the word-final sound [d] is not articulated).

Such departures from the canonical, dictionary pronunciation are one reason why DTW is frequently used to compute the distance between element *X* and element *Y*, where *X* and *Y* may be a word, a phrase, or even longer span of speech (e.g., a sentence).

Because pronunciation is inherently variable (without impacting intelligibility), the distance calculation is usually based on a large number (often hundreds or thousands) of signal parameters. Such complexity is then distilled into a computationally more tractable form using data-reduction methods such as feature selection (e.g., James et al., 2013, p. 203), principal component analysis (e.g., Jolliffe, 2002, or special-purpose neural networks such as autoencoders (Liou et al., 2014). The distance metric may include "high-level" features such as pronunciation error type, intonation, and other pitch properties (e.g., tone level and contour).

Using such comparative methods, Lee and Glass (2015) and Transparent Language's EveryVoice™ technology deploy

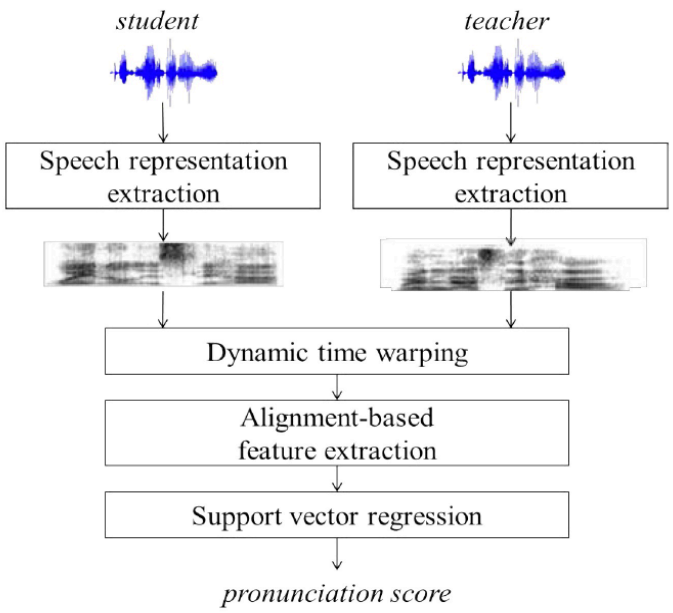

Figure 3. *An early version of Lee's pronunciation evaluation system (see Lee, 2016 for the complete system). After transforming the waveform into a speech representation, the system aligns the two utterances via DTW and then extracts alignment-based features from the aligned path and the distance matrix. A support vector regression analysis (a form of optimization) is used for predicting an overall pronunciation score. Reprinted from Lee and Glass (2013), with permission.*

DTW and DNNs to pinpoint pronunciation errors and offer remedial feedback. Lee's (2016) study is especially instructive. It uses DTW, alignment of the student's speech with a native speaker model, as well as machine learning to flag mispronunciations (**Figure 3** shows a simplified version of their system).

The more similar a student's speech is to the native-speaker model, the more successful the DTW-based alignment is likely to be. Instances where the alignment falters or shows anomalies are flagged as potential mispronunciations. The system also ascertains the specific form of error (substitution, deletion, or insertion). The benefit of this approach is its simplicity and adaptability to a broad variety of languages without the need for extensive customization.

Deep Linguistic Analysis

In classical ASR, a neural network is trained to recognize each of the dozens of different speech sounds (phones) in the phonological inventory of a language. Using a dictionary lookup, it's theoretically possible to represent thousands of words using just a few dozen symbols (which partially overlap with the 26 characters of the Roman alphabet). However, several dozen is still a large number with which to train a neural network, particularly when preceding and following phonetic contexts are considered. Such context-dependent models (Yu and Li, 2015) can number in the thousands, making neural network training especially challenging, especially for limited amounts of training material.

The training of the neural network comes in three basic forms (Bishop, 2006): (1) "supervised," in which the training data are explicitly labeled (as in **Figure 1**), preferably with time boundaries for segments and words; (2) "unsupervised," in which none of the training material is labeled; and (3) "semisupervised," in which a portion of the training is supervised and used for training on unlabeled data.

For many years, supervised training was the norm despite it being burdensome and expensive. Unsupervised and semisupervised training are becoming more popular as DNNs increase in sophistication.

Deep Learning Neural Networks

The architectures of deep neural networks are more complex (and powerful) than classical ANNs as the result of their enhanced connectivity across time and (acoustic) frequency. This power is often augmented with "long short-term memory" (LSTM; Schmidhuber, 2015) and "attention" (Chorowski et al., 2015) models, which further enhance performance. Goodfellow et al. (2016) is an excellent, comprehensive introduction to deep learning and related topics.

Neural networks trained for language instruction have an inherent advantage over those designed for speech dictation and search (e.g., Alexa, Google Voice, Siri) in that the lesson material is often scripted, with most of the words (and their sequence) known in advance. This knowledge makes it somewhat easier to infer which speech sounds have been spoken and in what order (through a pronunciation dictionary). However, this advantage is counterbalanced by the limited amount of data available to train CALL DNNs as well as the diverse (and often unusual) ways students pronounce foreign material.

DNNs have been used in the past where the amount of training material is less than ideal (for classical ANNs). However, even DNNs require a minimum amount of training data to succeed. For this reason, alternative strategies are used to compensate for the paucity of data. A popular approach is to reduce the number of training categories from several dozen to a handful by using a more compact representation such as articulatory-acoustic features (AFs; Stevens, 2002).

What are AFs? They are acoustic models that are based on how speech is produced by the articulatory apparatus. For

Figure 4. *"Pronunciation Practice" in Transparent Language's courseware for Spanish. The student's speech is analyzed and evaluated by deep learning neural network (DNN)-trained acoustic analyzers. Problematic portions are highlighted in* ***yellow****. The student can replay these and compare them with a native speaker. An overall score is shown on the meter (****right****). Published from Transparent Language, with permission.*

example, a phonetic segment can be decomposed into a cluster of acoustic "primitives" that fully distinguish it from other segments in the phonological inventory of the language. Among the most common AFs are "voicing," "manner of articulation," and "place of articulation." Voicing, a binary feature, refers to whether the vocal folds are vibrating (+) or not (–). For example, in the word pan, the initial consonant [p] is "unvoiced," whereas the vowel and following consonant [n] are "voiced." Manner of articulation indicates the mode of articulatory constriction impeding the flow of air through the vocal tract. Examples of manner of articulation categories are "vocalic" (e.g., the vowel in the word "pan"), "stop" (a.k.a. "plosive") consonant (e.g., [p] in "pan"), or nasal consonant (e.g., [n] in "pan"). Place of articulation refers to the locus of maximum vocal tract constriction. In our "pan" example, the initial consonant, [p], has a "bilabial" (both lips) anterior locus of articulation while the final consonant, [n], is produced with the tongue contacting the alveolar ridge (a central place of articulation). Each speech sound (and by extension, syllables and words) can be represented by an analogous set of articulatory features that varies over time.

In EveryVoice™, a native-speaker model based in part on AFs, is compared with the student's utterance by using DTW in concert with several distance metrics. Those speech sounds more than a certain distance from a native-speaker model are highlighted (**Figure 4**). The application also provides an "intelligibility score," which reflects a weighted average of student-native distances across the utterance.

In the future, comparative analyses will likely include a range of time frames (and linguistic levels) such as those associated with the syllable (ca. 200 ms), word (ca. 200-600 ms), and phrase (1-3 s; Greenberg, 1999). Human listeners usually require a second or more of continuous speech to reliably identify all words spoken (Pickett and Pollack, 1963). This extended listening interval is also often required for automatic systems to achieve optimum performance and may account for the recent popularity of "end-to-end" (ETE) and "sequence-to-sequence" (STS) processing in ASR systems (e.g., Prabhavalkar et al., 2017). ETE and STS systems integrate acoustic, pronunciation, and language models into a single, coherent process and so would likely improve the accuracy of ASR-based CALL.

Higher Level CALL Applications

Learning a foreign language involves more than speaking intelligibly. Grammar and vocabulary must also be mastered. Constant practice is key for fluency. Online courseware encourages the student to speak and listen in a broad assortment of realistic situations. In some applications, the student is prompted to respond with a relevant sentence or two. Software evaluates the response, progressing to more difficult material only after the student has demonstrated mastery of the current lesson.

A recent study using ASR goes beyond pronunciation to offer feedback on a variety of language skills, such as grammar and syntax, for students of Dutch (van Doremalen et al.,

2016). A European project, the "Spoken CALL Shared Task" (Baur et al., 2017), offers an illustration of how online evaluation and feedback may operate in the future. The competitive task was based on data collected from a speech-enabled online tool used to help young Swiss German teens practice skills in English conversation. Items were prompt-response pairs, where the prompt is a piece of German text and the response is a recorded English audio file. The task was to "accept" or "reject" responses that may or may not be grammatically and linguistically correct. The task involved more than conventional ASR because it also involved the ability to discern semantically and grammatically appropriate responses using natural language processing. The winning entry (from the University of Manchester, UK) used an ASR system trained with DNNs.

A somewhat different approach is used by the "Virtual Language Tutor" (Wik, 2011), which is an embodied conversational agent that can be spoken to and that, in turn, can talk back (via speech synthesis) to the student. The agent guides, encourages, and provides feedback for mastering a foreign language (initially, Swedish).

The Future of CALL

Several trends in language-learning software are worth noting. Most will likely be enabled through some form of deep learning, among which are the following:

Games

The app FluentU uses real-world video containing music, video, movie trailers, news, and inspiring talks and turns them into personalized language-learning lessons. LingoArcade, Mindsnacks, and DigitalDialects are just a few of the online sites for learning a foreign language using similar material, all within a game-based structure. Su et al. (2013) illustrate several ways to "gamify" dialogue learning for language learning.

Virtual Language Learning

Applications such as ImmerseMe and Mondly place the student in simulated, real-life scenarios, such as a bakery or restaurant, where language skills can be practiced in an engaging way. In these apps, ASR evaluates the student's responses and offers feedback.

Intelligent Language Tutors

Applications such as Duolingo are starting to use "chatbots" to interact with students on a variety of topics to enhance vocabulary and grammar skills. These bots are driven by a combination of ASR, natural language processing, and other forms of artificial intelligence to guide the student through language lessons in naturalistic settings.

Automatic Language Translation

A Defense Advanced Research Projects Agency (DARPA)-funded project, TransTac (Bach et al., 2007), was an early, albeit limited, attempt to provide automatic language translation in a handheld box (for deployment in the Middle East). Among the languages offered were Iraqi Arabic, and Dari. Waverly Labs sells the Pilot™, an earbud-enabled app that performs simultaneous translation in near real time for over a dozen languages. Google Translate offers the ability to translate from one language to another. Among the languages offered for paired translation are English, French, German, Italian, Portuguese, Russian, and Spanish. Google also provides an optical version (using a smartphone camera) that translates signs and other text into one's native language. Microsoft has demonstrated simultaneous translation between English and Mandarin Chinese powered by a DNN that can meld the speaker's voice characteristics with the translated speech. These applications are not especially useful (yet) because they lack the semantic precision and emotional nuance emblematic of human communication, so are best reserved for simple scenarios such as grocery shopping and sightseeing.

Speech Synthesis

The quality and naturalness of speech synthesis has greatly improved, largely due to the ability of DNNs to simulate voices with realism. Baidu's Deep Voice (Arik et al., 2017), Amazon's Polly, Microsoft's Cortana, and Google's Cloud Text-to-Speech (TTS) applications all use DNNs. Google offers TTS in a dozen languages. Deepmind's Wavenet (van den Oord et al., 2017) offers highly realistic synthesis for English and Japanese in multiple voices.

Voice Conversion

Speech synthesis has improved to the point where it is now possible to transform or meld the voice characteristics of one talker into another while preserving intelligibility. Current state-of-the-art systems (Toda et al., 2016) use a special-purpose Vocoder (e.g., STRAIGHT, Kawahara et al., 1999; WORLD, Morise et al., 2016) as the synthesis engine. Two of the more advanced voice conversion systems use DNNs, which include long short-term memory (LSTM)-based recurrent neural networks (Sun et al., 2015) or sequence-to-sequence learning (Miyoshi et al., 2017).

Brain Stimulation

Neurotechnology may play a role in foreign language curricula of the future. A $12 million DARPA grant to Johns Hopkins University (Baltimore, MD) and collaborating institutions explores whether the ability to learn a foreign language can be enhanced through modulating the activation of relevant parts of the auditory and speech areas of the brain through electrical stimulation of the vagus nerve (e.g., Engineer et al., 2015).

Brave New Language-Learning World

DNN-powered speech technology is likely to play an increasingly prominent role in language-learning curricula. As computational power increases and costs diminish, simulation technology will enable a student to inhabit a virtual language world for hours on end. This is likely the future of language instruction, for there is no better way to learn a foreign tongue than to reside in a community where it is spoken. Will it matter that the language community exists only virtually? Virtual reality gaming devices, such as the Oculus Rift™, will only improve over time, enhancing their educational potential. Indeed, language learning could become a "killer app" for educational VR. Stay tuned.

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BioSketch

Steven Greenberg worked on SRI's Autograder project in the early 1990s. More recently, he has collaborated on the development of Transparent Language's EveryVoice™ technology. He has been a visiting professor in the Center for Applied Hearing Research at the Technical University of Denmark, Kongens Lyngby, as well as a senior scientist and research faculty at the International Computer Science Institute in Berkeley, CA. He was a research professor in the Department of Neurophysiology, University of Wisconsin, Madison, and headed a speech laboratory in the Department of Linguistics, University of California-Berkeley. He is president of Silicon Speech, a consulting company based in northern California.

Archaeoacoustics: Re-Sounding Material Culture

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Archaeoacoustics probes the dynamical potential of archaeological materials, producing nuanced understandings of sonic communication, and re-sounding silenced places and objects.

Acoustical Experiments in Archaeological Settings

Acoustical First Principles in Practice: Echoes and Transmission Range

Atop a 150-meter-long, 3,000-year-old stone-and-earthen-mortar building, 20 to 40 meters higher than surrounding plazas, two Andean colleagues and I listened to cascading echoes produced via giant conch shell horns known in the Andes as *pututus* (see **Figure 1**). Riemann Ramírez, José Cruzado, and I were testing and documenting the performance of an archaeologically appropriate sound source at the UNESCO World Heritage site at Chavín de Huántar, Perú (available at acousticstoday.org/chavin), located at the center of a 400- to 500-meter-wide valley 3,180 meters above sea level. Our objective for this experiment, conducted in 2011, was to measure sound transmission via its return from landform features surrounding the site. Although we concurred that we perceived the echoes "swirling around from all directions," our mission that day was more than reporting subjective impressions. By recording the initial sound and returning echo sequence using a co-located audio recorder, along with the ambient conditions of temperature and humidity important to calculating the contextual speed of sound in air, I could make precise calculations in postsurvey data analyses regarding the distances of surfaces producing discrete echoes. Via this typical archaeoacoustical experiment, we confirmed that the closest rockface on the steep western hillside, known to locals as "Shallapa," produced discrete audible echoes with little signal distortion. The test also demonstrated that transmission of the sound of large *Strombus* pututus, which measure around 96 dB(A) at 1 meter, was effective to at least 1 kilometer away from the site because strong echoes returned 6 seconds later (Kolar et al., 2012, pp. 45-46). This range is consistent with undistorted and audible pututu sound transmission between the site and several archaeologically relevant landform features of the surrounding valley. Pututus such as these were excavated from the 1st millennium BCE architecture at Chavín and continue to be important throughout the Andes today. Therefore, our study not only provided dynamical specifics regarding pututus in the Chavín context but also measures extensible to the archaeology of societies such as the Inca empire that dominated South America 2,000 years later.

Archaeoacoustics: An Archaeological Science

Archaeoacoustics is a developing field that offers the acoustical community an opportunity to work across disciplines to explore the significance of sound throughout time and across cultures. Archaeoacoustical discoveries often begin with the documentation and mechanical explanation of sound effects or the experimental testing of what can be heard from where. However, *archaeology* is about putting such findings in human context.

Archaeology spans human time and is about understanding human experience through indirect evidence rather than direct accounts. From excavations of ar-

Figure 1. *Ancient sound-producing instruments. Shown are 2 examples of 3,000-year-old marine conch shell horns known as "pututus" excavated in 2001 as a cache of 20 at the Andean Formative ceremonial center at Chavín de Huántar, Perú. Photographs courtesy of José L. Cruzado Coronel (**left**) and John W. Rick (**right**). Programa de Investigación Arqueológica y Conservación Chavín de Huántar.*

chitectural ruins to examinations of recently abandoned places or discarded objects, archaeological discoveries stem from what archaeologists call *material culture*. An interdisciplinary and anthropological social science, archaeology reaches across fields to harness tools and expertise (Trigger, 2006). More than an application of acoustics to archaeology, archaeoacoustics mobilizes science, engineering, and humanities research to produce archaeological interpretation. Through methods including experimental tests, analytical models, and computational reconstructions, archaeoacousticians explore and demonstrate the dynamical potential and sensory implications of archaeological materials.

There are numerous and diverse examples of excellent archaeoacoustics research (e.g., see case study discussions in Scarre and Lawson, 2006), best recounted by the researchers themselves. Here, I offer an overview of experimental approaches to archaeoacoustics via firsthand accounts, including an interview with archaeoacoustics pioneer and Fellow of the Acoustical Society of America (ASA) David Lubman. An acoustical consultant, Lubman was awarded the Helmholtz-Rayleigh Interdisciplinary Silver Medal in Architectural Acoustics and Noise by the ASA in 2004 for work in noise and standards and for contributions to architectural and archeological acoustics (e.g., Lubman and Wetherill, 1985).

Archaeoacoustics in Practice: Multidisciplinary Research

An Interview with David Lubman

A common starting point in archaeoacoustics fieldwork has been the evaluation of location-based sound effects, especially in relation to historical accounts, mythological premises, and public and ceremonial architecture. Lubman (2016) has explored sound effects at the Maya site Chichén Itzá, México, since 1998. Lubman's approach to archaeoacoustics is exemplary in its melding of humanities perspectives, social science, and experimental and analytical acoustical methods. In his work, nonacoustical background research provides context for acoustical investigations. The importance of archaeological context to archaeoacoustical research should not be understated. Among the many secondary accounts of Lubman's research, some writers have devalued the anthropological information that Lubman considers in both research design and interpretation. Dismissal of nonacoustical forms of data that are culturally pertinent to an archaeoacoustical investigation demonstrates a basic misunderstanding of archaeology. Archaeologists interpret materials in cultural contexts and physical settings to create narratives about plausible aspects of past human life from the "things" and places that were important to individuals, groups, and societies (Wiley, 2002).

Lubman works independently of archaeological projects to explore the acoustics of places of persistent human interest. Lubman's method brings together knowledge from history, literature, and auditory science, yet the driving impetus is his multifaceted acoustical engineering expertise. In 2007, Lubman presented one such cross-disciplinary exploration, "The Acoustician's Tale: Acoustics at the Shrine of St. Werburgh" to the 42nd International Congress on Medieval Studies. In this research, Lubman looked to European literature and history to understand religious pilgrimages to shrine sites where saints would be *petitioned* (prayed to) through contact with their relics, such as the basis for Chaucer's 1387 *Canterbury Tales*. Such accounts serve in archaeology as *anthropological analogies* rather than as contextual

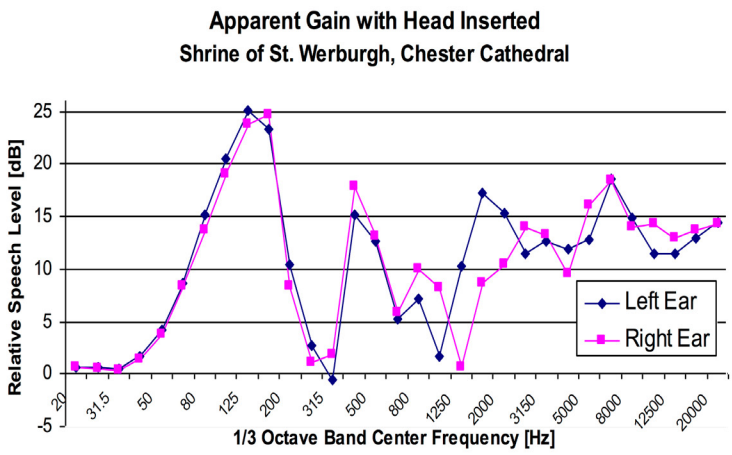

Figure 2. *David Lubman recorded binaural speech and calculated the apparent gain (**left**) produced by introducing one's head (as if to recite prayers) into the recesses of the St. Werburgh Shrine (**right**) at Chester Cathedral in 2003. Figure courtesy of David Lubman.*

evidence. Lubman recounts the study (Personal Communication, 2018):

"The unusual sound at the shrine of St. Werburgh, at Chester Cathedral (see chestercathedral.com) in western England, was brought to my attention in 2000 by the English architect Peter Howell and the architectural historian Julia Ionides of the Dog Rose Trust, a registered English charity. Peter and I visited the Shrine at Chester in July 2003. The shrine had been constructed, moved, rebuilt, damaged, and repaired, with these architectural changes traceable historically. I conducted an acoustical experiment to test functional questions about the role of sound in the *petitioning process*, the prayer requests a shrine visitor makes to the religious figure(s) represented in the shrine. The shrine is constructed with six recesses that can receive the head of a kneeling petitioner. In pre-Reformation times, prayers were spoken while petitioners knelt at the shrine with their heads in its recesses (**Figure 2, *right***). What did a petitioner hear? Did the shrine's acoustical architecture enhance the petitioner's experience? My acoustical experiment at the shrine sought to find the difference in speech quality and spectrum levels heard with one's head in the shrine versus one's head outside the shrine. I used head-worn binaural microphones to create a high-quality digital recording made with the talker's (my own) head first inside (see **Multimedia File 1** at acousticstoday.org/lubman-multimedia) and then outside the shrine recess (see **Multimedia File 2** at acousticstoday.org/lubman-multimedia), with the same vocal effort maintained in both recordings. I then produced a graph of the apparent gain with the head inside the shrine (**Figure 2, *left***), across third-octave bands in the hearing range, comparing the signal from both ears, that tracks how speech levels are greatly enhanced over the range of human hearing when one's head is located inside a shrine recess. From an interpretative perspective, recess acoustics elevate the petitioning event to "theater!" Within the shrine recesses, petitioners would hear their own voices reinforced, and they would thus be prompted to reduce voice level (in psychoacoustics, this is known as the *Lombard effect*). Inside the recesses, petitioners would be less aware of other sounds in the cathedral. The petitioners' voices are reverberated, creating a mysterious-sounding "reverberant halo," an effect that might seem like talking to another world. In this physical and religious context, the auditory percept of *proximity* may be interpreted as spiritual intimacy. My reconstructive experiment in re-creating petitioners' aural experience is a way of re-creating history, demonstrating how sensory experience is another way of knowing."

Lubman's study of the Shrine of St. Werburgh provides an empirical complement to historical archaeology, which draws heavily on written texts for experiential accounts. Lubman's experimental reconstruction produced a recorded demonstration, backed by acoustical metrics, for the architectural transformation of speech within the shrine recesses. Via archaeoacoustics, the effects that were once only possible to experience in person, in situ, can be demonstrated off-site via Lubman's audio recordings (see links above).The quantitative data from the archaeoacoustical experiment detail the amount of vocal enhancement specific to the experimenter, yet analysis of its frequency dependency enables the estimation of the shrine's acoustical effects for other talkers, thus making the research extensible to archaeological estimations. Archaeoacoustical scenarios that could be modeled using Lubman's data include charting the difference in acoustical feedback for people with different vocal ranges and characterizing a range

of potential experiences. Lubman's documentation and acoustical analysis of the sonic enhancement effect of medieval European shrine architecture demonstrates a physical basis for the spiritually transformative experience recounted in historical documents and elaborated in literature.

Sound as Archaeological Evidence: Archaeoacoustical Theory and Method

Disciplinary Background: Studying Sound in Archaeology

Because archaeology employs experts from many fields, the exploration of sound-related archaeological concerns by acousticians might seem a typical collaboration. However, acoustical science is a novel and infrequent addition to the archaeological toolkit, with sonic concerns typically given cursory mention if not ignored. Until recently (e.g., Scarre and Lawson, 2006), sound as a topic for archaeological inquiry was assumed common sense or relegated to musicologists, who primarily deal with nonsonic musical culture, such as textual and graphical representations of musical practices or the reconstruction of instruments and tuning systems. The habitual dismissal of sound as a topic for archaeological study may relate to the mismatch between ephemeral understandings of sound and the premise of contemporary archaeology. Archaeologists investigate human experience *indirectly*, inferring human actions on things and places from material evidence (such as "use-wear" marks on objects) rather than from direct accounts by individuals. Despite its material basis, archaeology often incorporates knowledge from the ethnographic work of anthropology or *ethnomusicology*, where testimonials and practices are recorded from living humans, or from the *narratives* that constitute written history, to form *analogical* or *corroborative* arguments. In practice, archaeological interpretation is a nuanced process of identifying and interrelating converging forms of evidence of human actions and related environmental factors.

Sensory Phenomena in Archaeology

Both archaeology and acoustics focus on materials. The inferential logic that transforms sound into archaeological material requires a discussion of mechanics and relationships. Such conceptualization is not unlike the logic that archaeologists use to trace the effects of human actions and environmental processes on cultural materials. However, studying sound and humans requires an examination of sensory, perceptual, and cognitive aspects of sonic experience. Human-produced and received sounds have physiological and psychological ramifications, studied via psychology in the direct study of living humans. In contrast, archaeology is about the indirect study of human life via materials. Although in recent decades, archaeology has taken an experiential turn (e.g., Shanks, 1992; Hamilakis, 2013), with growing discourse around sensory concerns (Day, 2013) and even incorporating cognitive neuroscience (Renfrew et al., 2009), such literature typically discusses sound from a philosophical rather than a scientific perspective.

Archaeoacousticians directly address the sensory implications of material archaeology and, although often reference psychoacoustical quantities, infrequently apply auditory scientific methodologies in detailed studies of archaeological sites or materials. My dissertation research leveraged experimental psychoacoustics to evaluate experiential implications of Chavín's interior acoustics, situating systematic auditory localization experiments within the archaeological architecture (Kolar, 2013). In these experiments, the sound stimulus was a recording of a site-excavated conch shell horn (a Chavín pututu), chosen for both its ecological validity to the archaeological context and its sonic characteristics of a noisy attack and tonal sustain. To facilitate a consistent stimulus across all combinations of source and listener locations, the pututu sound stimulus was recorded with a microphone located at the instrument bell and reproduced in the experiment through matching single-driver, directional loudspeakers (Meyer MM-4XP) calibrated to 96 dB(A) at 1 meter to approximate the sound level and directionality of these conch shell horns. **Figure 3** is an architectural illustration from survey data of one of the two Chavín galleries where the experiment took place, with a scaled 1.68-meter human figure depicting eight sequentially tested participant positions with facing directions (labeled "POS") and six separately sounding stimulus locations (labeled "SOURCE") where loudspeakers were directed away from nearest walls. The experiment produced data towards understanding how the waveguide-like architecture influences localization cues in this purported ritual environment (Kolar, 2013), research that initiated what I refer to as "sensory-spatial mapping" of the archaeological setting.

Reconstructing and Interpreting Archaeological Sound

Although this article features experimental archaeoacoustics research that explores extant architecture, instruments, and sites, some archaeoacoustics work is more theoretical, based on reconstructions using computational modeling techniques and dynamical estimations. For experimental observation, whether in situ or in models, sound must be generated via some form of vibratory excitation or a mod-

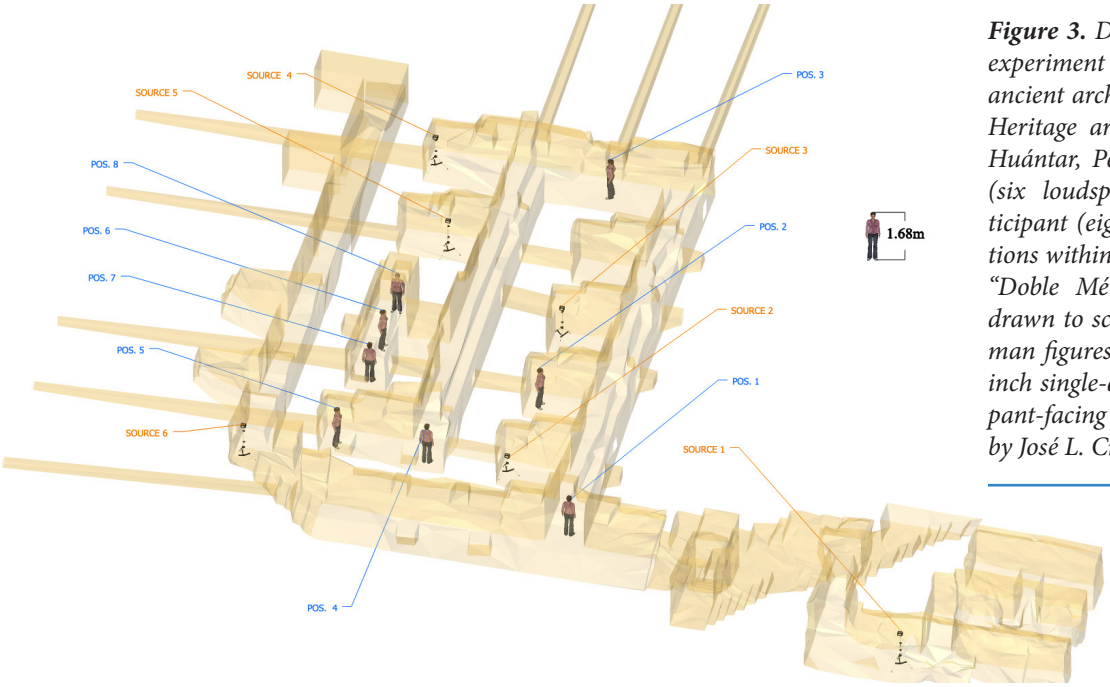

Figure 3. *Diagram of auditory localization experiment conducted in situ within the ancient architecture at the UNESCO World Heritage archaeological site at Chavín de Huántar, Perú. Shown are sound stimulus (six loudspeakers; "SOURCE") and participant (eight human figures; "POS") locations within the interior space known as the "Doble Ménsula (double-corbel) Gallery," drawn to scale as shown by 1.68-meter human figures. Directionality of calibrated 4–inch single-driver loudspeakers and participant-facing directions as drawn. Illustration by José L. Cruzado Coronel (Kolar, 2013).*

eled sound source. If archaeological sound must be reconstructed to be observed, is archaeoacoustics, therefore, a purely interpretative practice? Reconstruction and interpretation, although related, are not the same. The interpretative aspects of archaeoacoustical reconstruction depend on the way in which sound is produced as well as the choices of source and receiver locations that reenact human perspectives for contextual sound transmission.

Archaeoacoustical measurements made by exciting spatial or instrumental acoustics using an impulse (approximating a Dirac function) or a robust method for generating a spatial impulse response, such as the repeated exponential sinusoidal sweep technique developed and refined by Farina (2007), reveal archaeological acoustical features rather than reconstruct specific sounds. The impulse response can be thought of as a "spatial identifier," a composite acoustical feature set that reveals how the physical constituents of a space or instrument affect sound propagation. In contrast, human-performed acoustical test sounds, via artifact or replica instruments, are more interpretative, although the choice of particular instruments and the ways of playing them can be aligned with archaeological evidence. Reconstructive modeling and auralization of spatial and architectural acoustics likewise involve choosing sound sources and many other interpretative factors related to content, sound-making physics, and listener perspectives. Reconstructive interpretation, when informed by archaeological evidence, emphasizes the plausible rather than speculative.

Archaeoacoustical Interpretation in Archaeological Research

Archaeoacoustics produces assessments of the dynamical potential of archaeological materials, to support broader archaeological interpretation. The fieldwork and conservation program led by John Rick at the 3,000-year-old UNESCO World Heritage site at Chavín de Huántar, Perú, has invited and included archaeoacoustical collaboration since our project was formed at Stanford University in 2007. **Figure 4** shows several archaeoacoustical techniques employed in research at this well-preserved ceremonial complex that occupies about 14 hectares. In this research, converging forms of material cultural evidence support understandings of ancient communication (Kolar, 2017), including data from acoustical measurements of both site-excavated conch shell horns (Cook et al., 2010) and the well-preserved stone-and-earthen-mortar architecture. At Chavín, the only sound-producing instruments, either represented graphically (see **Figure 5**) or site excavated (see **Figure 1**), are the "Chavín *pututus*," marine shell horns made from the eastern Pacific giant conch *Strombus Lobatus galeatus*. Because no written texts are known from Chavín, we can only infer from material evidence, including extensive use-wear to the shells, that these instruments were performed at the site.

Pututus may have been performed in many places in and around the Chavín ceremonial complex during the 1st century BCE. Their performance physics in groups produces compelling effects for Chavín's ritual context, especially

Figure 4. *Since 2008, the author has adapted a variety of acoustical measurement techniques in fieldwork at archaeological sites including Chavín de Huántar, Perú (ccrma.stanford.edu/groups/chavin), using both loudspeaker-reproduced and human-performed sound sources, captured via multiple-microphone arrays and in-ear microphones. Photograph by José L. Cruzado Coronel.*

within the confines of interior architecture (Kolar, 2014; acousticstoday.org/pututus). However, converging forms of archeological evidence points to the performance of of pututus in and around the site's Circular Plaza. Alongside this 21-meter-diameter, semienclosed, countersunk plaza, the Chavín pututus were excavated in 2001 as a group, deposited along the walls of a small room. The plaza's decorated, relief-carved interior walls feature two known depictions of pututu performers (**Figure 5**), and several floor paving stones include fossil sea snails, the instruments' ancient ancestors. In 2009, acoustical impulse-response measurements were conducted in and around the partially intact Circular Plaza, within the Lanzón Gallery, the interior space to which it acoustically couples by way of three ducts. Repeated experiments using a precision loudspeaker and a spaced array of omnidirectional microphones through these ducts revealed that they are near-perfect filters for frequencies in the sounding-tone range of the Chavín pututus. The center duct between the interior gallery and exterior plaza, which is visibly aligned with the carved mouth of the *Lanzón*, a granite monolith historically reputed to be an "oracle" (**Figure 6**), further privileges pututu acoustics by emphasizing frequencies around 900 Hz (in the range of the instruments' third harmonic) that is an important timbral signifier (see **Figure 7**; Kolar et al., 2012).

Whether or not one concurs with the archaeological interpretation that suggests pututu performers could enact a metaphorical "line of speech" by sounding the instruments

Figure 5. *Relief-carved stone plaques lining the 21-meter-diameter Circular Plaza at Chavín de Huántar, Perú, depict figures holding conch shell horns (pututus) as if in performance. Photographs by José L. Cruzado Coronel and Miriam Kolar.*

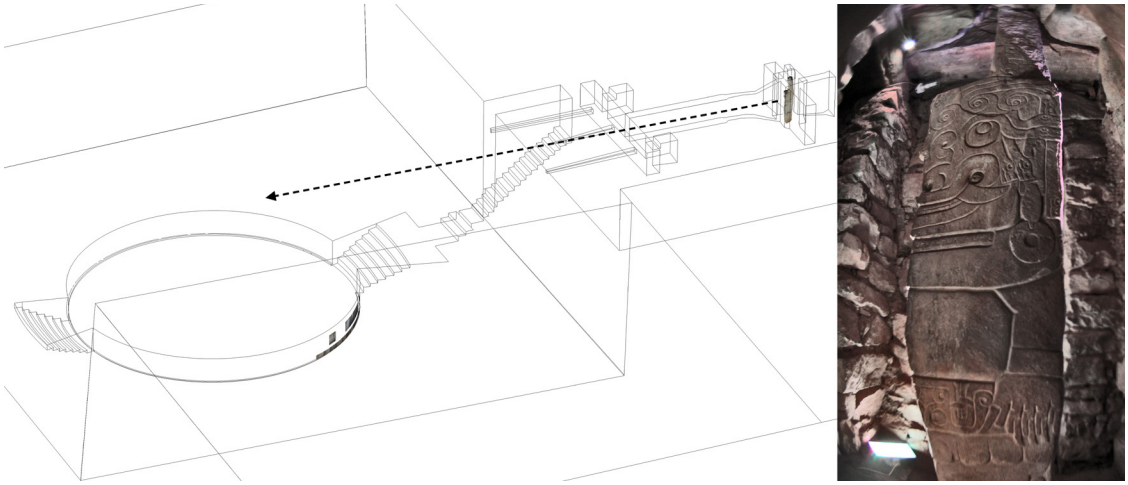

*Figure 6. Architectural reconstruction of Chavín's Lanzón Gallery and Circular Plaza (**left**) and the 4.5-meter granite monolith known as "the Lanzón" (**right**). Illustration and photograph by José L. Cruzado Coronel.*

between the Lanzón monolith and the Circular Plaza (see **Figure 6**; Kolar et al., 2012), repeated measurements have demonstrated that these ducts acoustically favor pututu sound and perceptibly filter out higher frequencies crucial to speech clarity, for example. Pututus would have been useful in transmitting sonic information between the access-restricted Lanzón Gallery, where the Lanzón "oracle" monolith (**Figure 6, *right***) is located, and the larger public gathering area outside, the Circular Plaza (**Figure 6, *left***). Whether or not the pututus would have been considered the voice of the oracle is an interpretative matter. From a physical dynamical perspective, we can assert that pututu sound transmission is facilitated architecturally between these spaces. In this research example, archaeoacoustics strengthens material archaeological associations by demonstrating dynamical context for the Chavín pututus within the ceremonial locus of Chavín's Circular Plaza. Architectural acoustical evidence, data from my team's acoustical study of the site-excavated pututus (Cook et al., 2010), and other archaeological information together support archaeological arguments for location-specific pututu performance at Chavín.

Archaeoacoustics and Music Archaeology

Likely due to the custom of identifying sound-producing instruments with music and an established scholarly path for musicological studies, the field of music archaeology precedes archaeoacoustics. Despite substantial attention to the acoustics of well-preserved amphitheaters, an area of archaeoacoustics dominated by architectural acoustical modeling research, European classical archaeology has emphasized musical concerns identified from texts and visual representations. Archaeological materials readily identified as "musical" are typically studied by music archaeologists, who employ musicological tools and methods concerned with the abstract, conceptual, structural, and performed aspects of music (its "culture") rather than sound (its "physics"), which has historically been the domain of musical acoustics. However, in archaeological practice, such culture-communication dichotomies are dissolving, and much as historical musicologists increasingly consider the acoustics of instruments and performance spaces, music archaeologists have begun to incorporate acoustical concerns.

Two recent studies led by scholars of art and architecture offer notable incorporations of archaeological acoustics, the Renaissance religious architectural study of Howard and Moretti (2009) and the multisensory exploration of Hagia Sophia in Byzantium by art historian Bissera Pentcheva (2010, pp. 45-56; demonstrated in this video available at acousticstoday.org/hagiasophia). Howard and Moretti's (2009) study included the reconstruction of musical performance practice in a dozen churches of Renaissance Venice, accompanied by audience surveys regarding perception of architectural acoustical attributes that were measured and modeled. Pentcheva's (2010) research considered the metaphorical value of sound in combination with light, human movement, and other elements of early Christian ritual in Constantinople. Historical musicologists and the choir Capella Romana worked with Pentcheva and Stanford musical acoustics colleagues to reconstruct period music as if performed within the 11-second reverberant setting of Hagia Sophia (heard on the video above).

For archaeological contexts including sound-producing instruments, it is difficult to avoid experimental and experiential engagements of archaeological materials. Making sound in places seems to have been a conscious human activity throughout time, as, for example, Morley (2003), Blake and Cross (2015), and Tomlinson (2015) among others have de-

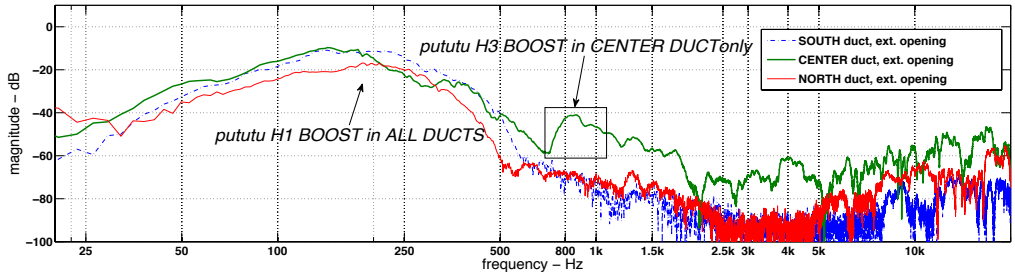

Figure 7. *Magnitude frequency measured at the exterior openings of the three ducts connecting Chavín's Lanzón Gallery with its Circular Plaza via the repeated sinusoidal-sweep impulse-response method. The sounding-tone range (H1) and articulation peak (H3) of site-excavated conch shell horns (pututus) are privileged by duct acoustics. Adapted from a diagram by Miriam Kolar and Jonathan S. Abel (Kolar et al., 2012, Figure 13).*

tailed. Indeed, sonic engagements with archaeological sites, whether or not musicological in purpose, have frequently stemmed from reconstructive soundings (often hand claps, footsteps, or whistles) as, for example, archaeoacoustics pioneers Paul Devereux (2001), Iegor Reznikoff (2006), Wayne Van Kirk, David Lubman, and rock art specialist Steven Waller have recounted in professional venues and in the popular press, among work by others too numerous to list here. *Acoustics Today* previously featured the work of Jelle Atema (2014), Professor Emeritus of Biology, Boston University, and Adjunct Scientist, Woods Hole Oceanographic Institution, MA, a flutist who studied with renowned performer Jean-Pierre Rampal. Atema has innovated the experimental reconstruction, performance, and organological exploration of flute technology, offering a comprehensive, physics-based perspective on ancient music making.

Although cross-disciplinary expertise is a hallmark of individual archaeoacousticians, collaborations across multiple fields drive unprecedented explorations of ancient sonics, which often result in formal musical performances for audiences. In 1992, musical acoustician Murray Campbell, musicologist John Purser, archaeologist Fraser Hunter, silversmith John Creed, and musician John Kenny began a multidisciplinary archaeomusicological reconstruction of the *carnyx*, a Celtic brass instrument based on fragments excavated in northeastern Scotland (Campbell and Kenny, 2012; acousticstoday.org/carnyx). Their collaboration has produced numerous archaeological engagements, including concert presentations of the carnyx in venues such as the 2018 Experimental Music Archaeology Symposium at the State Archaeology Museum in Brandenburg, Germany. Musicians such as Swiss trombonist Michel Flury have explored archaeological contexts to develop new musical interpretations on replicas of ancient instruments, such as Flury's series of Chavín-inspired performances with modern pututus that were featured in a local concert in that Andean town, followed by music for an international exhibition by the Museum Rietberg in Zurich, Switzerland, and continuing in current work (Flury's *Klanginstallation Chavín* available at vimeo.com/245501948). Beyond performance practice, music archaeologists are increasingly incorporating acoustical concerns and methods to characterize and contextualize musical materials, especially for artifact instruments of sound production that can be played or convincingly reconstructed (Both, 2009).

Mapping the Potential for Sonic Communication

Following the premise of sound as a near-universal means for human communication, archaeoacoustics is frequently concerned with establishing the plausibility of what can be heard and from where, dependent not only on acoustical science but also information from site archaeology. Archaeological context includes considerations about who would be hearing what sonic material, under what environmental conditions, and in what social or political settings. Archaeoacoustical studies frequently seek to test interpretative or historical claims as well as provide experimental evidence for sonic dynamics not reported or considered by others. For comparison and contrast with my initial discussion of the Chavín pututu echo study and to show how archaeoacoustical tools and methods can be adapted across archaeological contexts, I offer an example of an outdoor archaeoacoustical survey that also employed a *Strombus* pututu as one of several sound sources. To produce empirical data on site-specific sound transmission as well as test claims from many archaeological and historical accounts regarding the role of sound and architecture in Inca governance, archaeologist

Figure 8. *Acoustical survey on and around the central platform ("ushnu") at Inca Huánuco Pampa, Perú. Photograph courtesy of Miriam A. Kolar.*

R. Alan Covey, Andean experimentalist José Cruzado, and I designed and conducted an acoustical survey at the large Inca administrative city Huánuco Pampa. This imperial complex, active in the early 16th century, occupies a remote, high-Andean *pampa* (plain) 3,800 meters above sea level. Site architecture is organized around a plaza measuring 550 × 350 meters (19 hectares) with a raised central platform of 32.5 × 48 meters (see **Figure 8**; Kolar et al., 2018).

Conch shell horns figure prominently among sound-producing instruments mentioned in Spanish colonial accounts of the Inca empire, where they were known as long-distance communication devices carried by *chasqui* messengers. In the acoustical study at Huánuco Pampa, we used a *Strombus* pututu as one of a sequence of archaeologically appropriate instrument types to cross-compare the effects of frequency and production mechanism across mapped survey points. To provide a standard reference, we employed an electroacoustical test signal that is preferred for architectural acoustical measurements to produce impulse responses, which we also generated manually via a handheld percussion instrument (wooden clappers). In the broad Andean plain where Huánuco Pampa is located, simultaneously surveying colleagues reported hearing our tests in distant site sectors. Extrapolating our measured sound levels over the site map demonstrated the likely audibility of pututus to its perimeters (which extends 1.7 kilometers from the central platform), consistent with other data on pututu sound transmission. Postsurvey analyses of the recorded audio suggested that the particular frequency range of large Andean pututus (centered around 300 Hz), in combination with typical ambient daytime conditions in the central Andes (low humidity and moderate temperatures), makes them practically immune to wind shear, which is one of the environmental characteristics of high-altitude Andean sites, especially in the late morning through afternoon (Kolar et al., 2018). Theory-backed acoustical experimentation thus supports cultural evidence linking these instruments to political power in the Andes from the present back to the Inca (approximately 13th to 16th centuries CE) and as early as Chavín (1st millennium BCE).

Archaeoacoustics: Acoustical Science in the Service of Archaeology

Working at a new scientific frontier, archaeoacousticians responsively adapt acoustical science methods to archaeological research. An archaeological science, archaeoacoustics enables specific characterization of sound-related matters and methods for evaluating the extensibility of findings from one context to others or generalizing findings to a broader archaeological interpretation. Archaeoacoustical research worldwide has demonstrated the feasibility of adapting acoustical theory and methods to diverse archaeological sites and materials. Archaeoacousticians re-sound silent traces of past life, bringing the past into the sensory presence. This unique combination of science and humanities research provides novel opportunities for thinking and working across disciplines. Archeoacoustics connects the human experience across time and geography.

Acknowledgments

Many thanks to archaeoacoustician David Lubman for the interview given here and for his generous invitations to participate and collaborate in archaeoacoustical discourse (dlubman@dlacoustics.com). Arthur Popper's keen editorial sensibilities greatly enhanced the direction of this text.

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BioSketch

Miriam Kolar's cultural acoustics research leverages acoustics and psychoacoustics to study sound in cultural contexts. Since 2008, she has led archaeoacoustics research at the UNESCO World Heritage site Chavín de Huántar, Perú, where her methodological innovations include on-site auditory localization experiments to evaluate experiential implications of archaeological architectural acoustics. Recently a Weatherhead Fellow at the School for Advanced Research (SAR), Santa Fe, NM, Kolar received her PhD as a Stanford Interdisciplinary Graduate Fellow at the Center for Computer Research in Music and Acoustics (CCRMA), Stanford University, Stanford, CA. Prior to her doctoral study, Kolar engineered concert sound design and location recordings, and directed the CalArts under-graduate music technology program.

From Sputnik to SpaceX®: 60 Years of Rocket Launch Acoustics

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The field of rocket launch noise is 60 years old and has a lot to celebrate!

In the Beginning

At 7.28 pm (GMT) on October 4, 1957, the Soviet Union launched a 58-cm-diameter polished metal sphere into an elliptical Earth orbit at 29,000 kilometers per hour (kph), 800 km above the Earth's surface. The satellite was equipped with two pairs of external radio antennae that broadcast pulses from a one-watt, 3.5-kg radio-transmitting unit inside the sphere. Signals were emitted on two frequencies, 20.005 MHz and 40.002 MHz, spending 0.3 second at each (see bit.ly/2fSYJNl to hear these transmissions). The successful launch of Sputnik took the world by surprise at a time when the USSR and the United States were both desperately seeking to establish dominance over the new frontier, space.

Sputnik 1, where sputnik means "something that is traveling (the satellite) with a traveler (the Earth)," was the world's first artificial satellite. However, the language of rockets, the launch vehicles for satellites, goes back centuries. According to the Merriam-Webster Dictionary New Edition (2016), the first known use of the word rocket was in 1611. Deriving from the Italian word *"rocchetta"* meaning distaff, a rod used in spinning or weaving, it is defined in the Merriam-Webster Dictionary New Edition (2016) as "*a jet engine that operates on the same principle as the firework rocket, consists essentially of a combustion chamber and an exhaust nozzle, carries either liquid or solid propellants which provide the fuel and oxygen needed for combustion and thus make the engine independent of the oxygen of the air, and is used especially for the propulsion of a missile (such as a bomb or shell) or a vehicle (such as an airplane)*." Although we think of rockets as relatively new inventions, the first device to use the basic principles of rocket propulsion was probably a wooden bird, invented somewhere around 400 BC and propelled by escaping steam using the law of action-reaction (see bit.ly/2BNOi7F).

During the early days of the Space Race, the risk of rockets exploding during or just after liftoff was high and the technical focus was on improving rocket reliability. As the prospect of launch failure diminished (Martinez-Val and Perez, 2009), attention turned to other aspects of the liftoff process. At the time, very little was known about the acoustics of rocket launches, and at 300 m from the launch pad, peak levels could reach as much as 200 dB during liftoff (as far as 3 km from the launch pad, peaks of 140 dB, which is the threshold of pain in humans, were being observed). Such high acoustic loads are a source of vibration, and vibroacoustic interaction can critically affect correct operation of a rocket and its environs. Thus it was quickly realized that even a small decrease in noise could result in a significant reduction in both risk and cost, and the discipline of Launch Vehicle Acoustics was born.

And Then Came Sputnik...

Many important breakthroughs in rocket design were made between the 16th and 19th centuries. For example, the concept of a vehicle with multiple stages that

Figure 1. *Dr. Robert H. Goddard before the flight of his first liquid-propelled rocket in Auburn, MA, on March 16, 1926. Photo available at go.nasa.gov/2MPRSCD.*

could be used to lift rockets to higher altitudes, the so-called step rocket, was initially developed in the 16th century by a firework maker for enhancing firework displays (see bit.ly/2Ln7TLH). Its principles are still fundamental to the launch of orbital vehicles today. A significant breakthrough came in the 17th century when Sir Isaac Newton developed the scientific foundations for modern rocketry. His three laws governing physical motion explained how and why rockets work and could be used to inform future rocket design. However, improvements in rocket accuracy were not achieved for another two hundred years, until the technique of spin stabilization was developed. In spin stabilization, the exhaust gases strike small vanes, causing the rocket to spin as it travels. This is the same principle that bullets use today.

Still, modern rocketry belongs to the 20th century. Although early rockets were propelled by solid fuels, modern scientists thought (Tsiolkovsky, 1903; Goddard, 2002) that they could achieve greater speed and range by using liquid fuel. American rocketry pioneer Dr. Robert H. Goddard (see **Figure 1**) achieved the first successful liquid-propelled rocket flight in Auburn, MA, on March 16, 1926 (see bit.ly/2Lr9FeI for video of the first liquid-fueled rocket launch). The outbreak of World War II caused attention to shift almost exclusively to the development of rockets for use as weapons. They became such important elements of warfare that had research advanced more quickly on Germany's V-2 rocket, designed by Wernher Von Braun, the course of the war would almost certainly have been significantly changed. Similarly, it was the military as well as the scientific uses of rocket technology that made Sputnik such a historic event. Its successful launch caused great anxiety for the Americans who feared a widening technological gap between the two so-called superpowers, the United States and USSR.

In December 1957, American scientists made their first attempt to launch a satellite into orbit. The Vanguard TV3 ignited and began to rise but immediately lost thrust and fell back to the launch pad (see bit.ly/1bvmJxp to view the unsuccessful Vanguard TV3 launch). Finally, in early 1958, the United States successfully launched its first satellite, Explorer 1, which returned data for nearly four months and which remained in orbit until 1970 (see bit.ly/2P5dQPy that shows the US (National Aeronautics and Space Administration [NASA]) Space Explorations of 1958, including the first five US satellites). Following Explorer 1, satellite technology developed quickly and by the mid-1960s, satellites were prevalent and were being widely used for digital telecommunications. To date, more than 6,600 satellites have been launched from 40 countries. There are currently approximately 3,600 in orbit, 1,000 of which are operational.

And We Have Liftoff...

Rocket "launch" is the liftoff phase in a rocket's flight. Orbital launch vehicles, rockets that are capable of placing payloads into or beyond Earth orbit, typically lift off vertically (or near vertically) before progressively leaning over as they use gravity to steer the vehicle onto the required trajectory in order to exit Earth's atmosphere. Once rockets have completed the transonic climb phase through the atmosphere, they reach the desired altitude by angling slightly below the horizontal. This maneuver also increases their horizontal speed until it reaches orbital speed, at which point the engine cuts out. Because single-stage orbital rockets require an excessive amount of fuel, all current rockets are multistage vehicles, meaning that they jettison hardware (stages) on the way to orbit. The jettisoned stages are either lost or recovered and reused as in SpaceX's recently developed Falcon rockets.

The launch environment, which typically lasts only a few minutes, is the most severe dynamic environment that a spacecraft will endure during its normal life (Martinez-Val

and Perez, 2009). The acoustic environment of a launching rocket is two-phase. During hold-down, which lasts a matter of seconds, the first stage engines are firing and building thrust, but the rocket is restrained by the transporter erector launcher (TEL). The second phase is entered once the TEL releases and the rocket lifts off, initially moving very slowly. During both phases, a dynamic load is produced on the surrounding infrastructure and personnel by sound pressure waves that fluctuate and generate structural vibrations that, if they are strong enough or at the "right" frequency, can cause damage or injury (Hess et al., 1957).

Since the late 1950s, engineers have been concerned about the acoustic environment generated by rockets. During the development of the Saturn V launch vehicles, still the tallest, heaviest, and most powerful rockets launched to date (see bit.ly/2P7N09E for the Apollo 8 Saturn V launch), there was a great deal of concern about the acoustic impact their launch from Cape Canaveral would create. A novel solution was suggested, namely, moving the launch site offshore to a remote structure built in a deepwater location. Three radar facilities off the east coast of Texas (the Texas Towers) had already been used during the Cold War as surveillance stations, and it was suggested that one tower be repurposed for use as a launch pad. However, after a 1961 storm destroyed one of the towers, the idea was abandoned (Teitel, 2016).

The eventual ground launch of NASA's Saturn V rocket was, at 204 dB, one of the loudest sounds ever recorded. This focused attention on improving predictions of liftoff noise so as to affect rocket design and thereby reduce damage from launch-generated noise (Guest and Jones, 1967). However, the upcoming launches of SpaceX's Interplanetary Transportation System (ITS) and NASA's Space Launch System (SLS), extremely large rockets with big acoustic impacts, are likely to generate renewed interest in offshore launches.

Launch Vehicle Acoustics: An Overview

Rocket launches generate a significant amount of acoustic energy. The primary source of rocket noise is due to the high jet exhaust velocity required to boost the launch vehicle during takeoff. Shock waves are formed by the collision of the supersonic exhaust plume with the ambient air, and the acoustic intensity of these waves depends primarily on both the size of the rocket and its exhaust velocity. Typical near-field peak noise levels are around 170-200 dB and are concentrated in the low- to midfrequency range, namely 2 Hz to 20 kHz. This is exactly the range where the transmitted energy and power can cause damage to buildings and humans (Teitel, 2016).

Turbulent boundary layer excitation, separated flows, and wake flows also contribute to an extremely inhospitable acoustic environment that can cause structural vibrations during the climb through the atmosphere. Once the vehicle is supersonic, the rocket exhaust noise becomes less than the turbulent flow noise excitation. When stages separate, pyroshocks (the transient dynamic structural shock that occurs when an explosion or impact takes place on a structure) occur, causing additional vibration problems. However, it is the launch phase (characterized as a random, nonstationary, short-duration transient) that is the most problematic in terms of generating a potentially damaging vibroacoustic profile (Arenas and Margasahayam, 2006).

There are three types of supersonic jet noise: turbulent mixing noise (TMN) and two types of shock-associated noise (SAN): broadband shock-associated noise (BBSAN) and discrete screech tones (Allgood et al., 2014). TMN is always present and is generated by the large-scale turbulence structures/instability waves of the jet flow. However, the two types of SAN only occur in jets where there is a mismatch between the pressure at the jet exit and the ambient pressure (so-called imperfectly expanded jets). In this case, pressure equalization takes place through a series of compression and expansion cells or shock cells that form in the jet plume. BBSAN is then caused by the interaction of turbulence in the jet shear layer with this shock-cell structure and is primarily directed back toward the jet nozzle. Under the right conditions, BBSAN can also lead to the formation of narrowband tones, known as screech tones.

Not surprisingly, accurate prediction of the overall sound and vibration fields emitted by a rocket jet based on the rocket engine design is extremely difficult because it comprises several different, complicated noise-generating mechanisms (see **Figure 2**) and requires a detailed knowledge of the associated thermodynamics, aerodynamics, and acoustics (Koudriavtsev et al., 2004). Further complications occur when the effects of the launch vehicle and payload, launch pad design, and surrounding infrastructure are taken into account. Consequently, much rocket launch noise work to date has focused on noise mitigation, on experimental work, or the development of models that combine experimental data and theoretical assumptions.

Noise Mitigation

In many engineering applications, noise mitigation can be achieved by the control of vibration boundaries and unsteady flow phenomenon. Such techniques can be divided

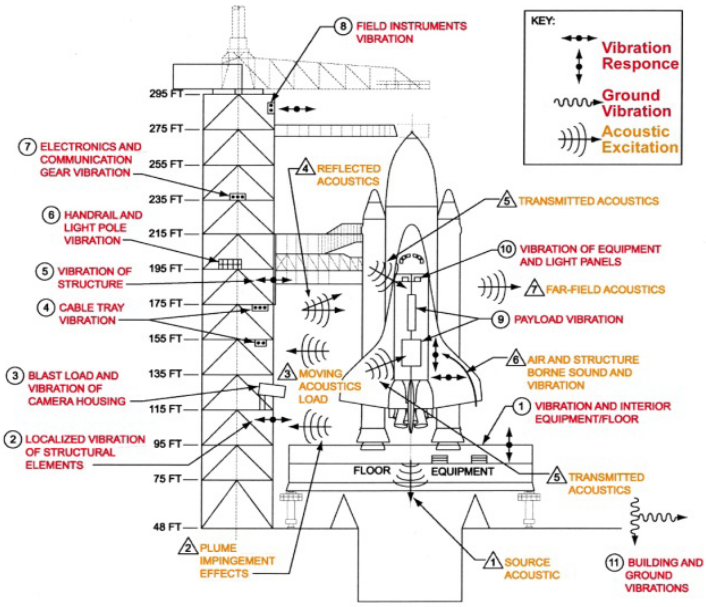

Figure 2. *Rocket launch-induced noise and related vibration-induced response of structures.* ***Numbers in circles*** *are related to those effects of vibration response that need to be considered as potential problems during the operation of the shuttle. These include structure-borne wave transmission.* ***Numbers in triangles*** *are related to the acoustic radiation, such as airborne sound waves. Reproduced from Arenas and Margasahayam (2006, Figure 8).*

Figure 3. *Water from an acoustic suppression system test soaks the mobile launcher platform at the Kennedy Space Center Pad 39A with 350,000 gallons of water. Photo available at bit.ly/2P1W93e.*

into active noise control, which results in achieving sound reduction in real-time using a power source, and passive noise control, which incorporates sound reducing measures into the original system design or retrofits them. Passive treatments are most commonly used to mitigate rocket launch noise.

Water-based acoustic suppression systems are commonly used on launch pads (see **Figure 3**), where they offer typical noise reductions of 3-5 dB (depending on frequency) in the overall sound pressure level at most frequencies of interest (Krothapalli et al., 2003; Norum, 2004; Houston et al., 2015). Interestingly, the technology on which these suppression systems are based was originally developed to help submarines avoid detection. Naval engineers designed the exhaust of submarine engines to emit bubbles, which have the ability to absorb an amazing amount of sound. As sound waves encounter the bubbles, the bubbles compress and convert the acoustic energy into heat, thereby shrouding both the noise emitted by the submarine, and incoming sonar waves.

In a water-based launch pad acoustic suppression system, water molecules sprayed into the air begin to vibrate on contact with a sound wave, converting the acoustic energy into heat. Additionally, any air bubbles present in the water will be compressed by sound waves, again converting the sound energy into heat energy. At the same time, below-deck (that is, under the launch pad) systems inject water into the exhaust plume with the aim of reducing far-field noise by more rapid dispersion of the rocket exhaust (Allgood et al., 2014). Moreover, above-deck systems, so-called rainbirds, inject water around the top of the pad as well as into the plume (Houston et al., 2015) that, in addition to suppressing the noise, helps cool the launch pad and environs. Care must be taken, however, not to deluge the pad, degrade materials and structures (Pico et al., 2016), or adversely affect performance of the diffuser (Allgood et al., 2014), which is a device used during sea-level rocket tests to simulate the effects of altitude. Water injection helps to reduce the SAN (Norum, 2004), and the extent of the reduction depends on where in the plume the water is injected and how great the injection pressure is (Gely et al., 2000, 2005; Lambare, 2016). To achieve any significant noise reduction, the quantity of water injected must be at least three times the jet flow rate; for the new acoustic suppression system on the mobile launcher platform at the Kennedy Space Center (KSC) Pad 39A, this means that the water flow rate exceeds 900,000 gallons a minute at liftoff. See bit.ly/2o9FEa7 for a recent test of the water suppression system (using about 450,000 gallons of water) at the KSC Pad 39B, from which the SLS will launch.

A key component of any launch pad is a flame deflector (FD), which is a trench used to channel the rocket exhaust away from the pad (see **Figure 4**). Flame deflectors are generally not specifically designed for acoustic purposes but nonetheless can have an important effect on the noise. Although the impingement of the plume on the FD generates noise that propagates away from the vehicle, the unsteadiness of the plume flowing along the FD emits the dominant noise that is directed toward the vehicle. As a consequence, factors such as trench cover and shape have a significant effect on the ability of the deflector to reduce noise (Gely et

Figure 4. The Antares launch pad at the Wallops Flight Facility, Virginia. The rocket is attached to the pad by the rocket launch mount, which is the large white ring that can be seen in the ***middle*** *of the figure. The (square) rocket flame trench exit is shown at* ***right****.*

al., 2005; Pico et al., 2016). Preliminary work (Tsutsumi et al., 2009) has indicated the possibility of substantial noise reduction, particularly if the initial inclination of the FD is steep and it is covered. The longer the trench, the greater the noise reduction (Gely et al., 2000).

Rocket nozzle configuration and shape also impact launch noise emission (Humphrey, 1957; Viswanathan et al., 2012), and tailored nozzles can provide a reduction in the directional noise by providing a low-speed layer around the outside of the primary jet that partially blocks sound transmission. This layer can be further modified by the use of wedges, pairs of vanes, and flaps (Viswanathan et al., 2012). Finally, flat concrete (reflecting) surfaces are predominant on launch pads, and recent studies have indicated that the inclusion of perforations in these surfaces is effective at reducing noise (Natarajan and Venkatakrishnan, 2016).

To control the vibration levels on launch structures, their dynamic characteristics need to be thoroughly understood, and a significant amount of recent work has focused on this (Caimi and Margasahayam, 1997; Margasahayam et al., 2002). Whereas previous pad configurations have been designed based on reducing liftoff peak acoustic load, Caimi and Margasahayam's (1997) work indicates that the duration of plume impingement is a far more damaging and crucial design parameter. However, it should be noted that the feasibility of utilizing such modifications in practical launch pad design still remains to be determined.

Theoretical Work and Scale-Model Experiments

Noise mitigation techniques have been fairly successful and in some cases decrease peak acoustic levels by up to 5 dB. However, to achieve further reductions, much greater understanding of the mechanisms by which rocket launch noise is generated and propagated is necessary. Although the importance of acoustic loading in causing structural failure has been known for 60 years (e.g., Hess et al., 1957), only relatively recently have significant advances in sensors, data acquisition, and processing techniques, along with huge improvements in numerical simulation ability, allowed the measurement and prediction of launch noise with any degree of accuracy.

The main issue in accurately predicting rocket launch noise is determining the relationship between the aerodynamic characteristics of the flow and the spatial characteristics of the sound field. Most rocket noise models are semiempirical and based on the classic NASA SP-8072 methodology (Eldred and Jones, 1971) in which the rocket plume is the primary noise source throughout launch. A major problem with this method is that it is not consistent with Lighthill's (1952, 1954) generally accepted jet noise theory because the initial approach to the aerodynamics was far too simplistic. Nevertheless, modified versions of the NASA model continue to be used. Revisions typically focus on improving the estimate of the laminar core length (Varnier, 2001) or on amending the acoustic efficiency by a factor that significantly improves the fit to the experimental data while accounting for the launch pad and FD geometry as well as for shielding (Plotkin et al., 2009).

Other empirical or semiempirical methods based on historical data, engineering judgment, and/or acoustic measurements have also been used extensively (Arenas and Margasahayam, 2006; Fukuda et al., 2009). For example, data were recently collected by the Japan Aerospace Exploration Agency (JAXA) during two static-firing tests of a solid rocket motor. The data were then compared with the results of the classical NASA SP-8072 empirical prediction method and a computational fluid dynamics (CFD) calculation (Herting et al, 1971). The former overestimated the sound pressure level at certain angles from the jet axis, although the prediction at other angles was reasonable. The CFD model was effective for prediction of both the near- and far-field acoustic profiles.

Once the acoustic load generated by liftoff has been predicted, it is then used to predict internal vibration responses of the vehicle, its payload, and the launch pad. Due to the

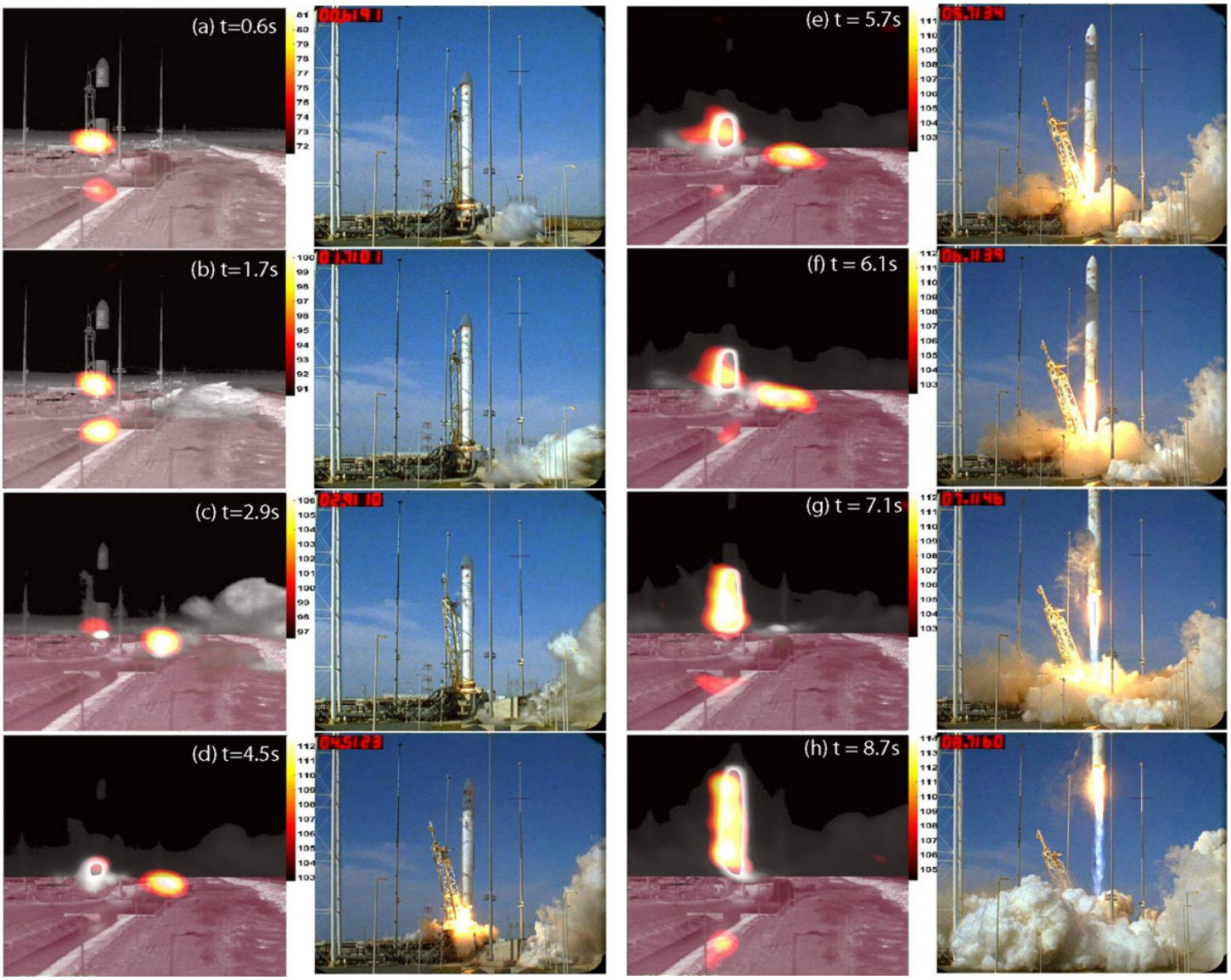

***Figure 5. Left, a-h**: noise maps superimposed on frames from an infrared camera, conventional beamform at 2 kHz; **right**: corresponding frames from a high-speed camera. Reproduced from Panda et al. (2013, Figure 11). See text for further explanation.*

obvious difficulties in full-scale experiments, launch vehicle design is heavily reliant on scale-model testing, where the results can prove invaluable (Bies and Franken, 1961). A recent comparison between full-scale and model-scale data shows good correlation, indicating that scale data can be used with confidence (Giacomoni and Kenny, 2016). Indeed, a source localization and reconstruction technique has lately been employed to successfully analyze wall pressure measurements on a model launch vehicle (Casalino et al., 2012). Similarly, aerovibroacoustic methods are currently being developed for predicting the response of a rocket to the intense acoustic environment inside the nose cone used to protect the payload (Tsutsumi et al., 2016). Such research is of great importance in rocket design.

Recent efforts have also focused on collecting acoustic data from static-fire rocket tests in an attempt to characterize the full-scale rocket plume noise environment. The acoustic temperature has been measured in a rocket noise field and found to contribute significantly to the total temperature variations (Giraud et al., 2010). Near-field vector intensity measurements on a model rocket motor indicate that as the frequency increases, the dominant source region contracts and moves upstream, with peak directivity occurring at greater angles from the plume axis (Gee et al., 2010). The noise source, conventionally assumed to be the rocket plume, is known to be directional and distributed and can be modeled by line arrays of monopoles that mimic the partially coherent nature of jet noise (Morgan et al., 2012). Recent work has shown that including source correlation and atmospheric turbulence in the model improves the predictions (Gee et al., 2014).

The first beamforming experiment conducted during an actual launch, in which microphones were placed on the rocket itself, confirmed the source distribution found dur-

ing a model-scale static-fire test (Gely et al., 2000). However, the first time a ground-based beamforming experiment was conducted during an actual launch was not until the 2013 launch of Orbital ATK Antares rocket from the NASA Wallops Flight Facility in Virginia (Panda et al., 2013). Results from these experiments provided unprecedented and unexpected insights into rocket launch noise sources (see **Figure 5**). In contrast to previous static-fire tests, they indicate that the primary noise source changes with time and that the source distribution is actually very different from the traditional model assumption of the plume as the primary noise source throughout launch (Eldred and Jones, 1971).

Figure 5, *left*, shows the sound sources, with lighter colors indicating higher noise levels, as identified by the beamforming experiments. **Figure 5, *right***, shows selected frames from a high-speed camera and are courtesy of the NASA KSC imaging group. From **Figure 5, a** and **b**, it is clear that during initial engine ignition, the primary noise source (***left, bright yellow/white areas***) was the launch mount and its accompanying ground reflection. However, once the engines come to full power during hold-down, the hot exhaust plume exits the deflector and the FD exit becomes the primary source (see **Figure 5, c** and **d**). This effect is mitigated to some degree by the duct water. The TEL then releases the launch vehicle and pitches away from it while the launch vehicle simultaneously fires at a slight angle to the vertical away from it. This so-called "TEL avoidance maneuver" causes the hot exhaust plume to spill out from the FD inlet and spread across the launch pad, causing a large area on the surface of the pad to become a loud, distributed acoustic source, in addition to the FD exit (see **Figure 5, e** and **f**). Thus, contrary to assumptions made in many traditional rocket noise models, it is not until the plume finally emerges fully from the duct (see **Figure 5, g** and **h**) that it becomes the primary noise source.

These data indicate that, contrary to the traditional model assumptions, a thorough understanding of the changes in acoustic source location with launch phase (time) is of fundamental importance in accurate launch noise modeling. For example, within the Antares flame trench, the Coanda effect is present. In the Coanda effect, a jet of fluid passing over a curved (Coanda) surface bends to follow that surface, simultaneously entraining large amounts of air as it does so. Because this flame deflector is the primary noise source as the engines come to full power (see **Figure 5, c, d** and **e**), it may prove to be a significant source of launch noise. Thus, recent work focuses on applying results obtained previously concerning turbulent Coanda jet flows (Lubert, 2008, 2017) to modeling the noise generated within this deflector.

The Future

Expendable launch vehicles have been used in the vast majority of the approximately 5,700 launch attempts since Sputnik. Now, however, the trend is for reusable rockets (Klotz, 2017). Indeed, the US Air Force and NASA, the two biggest customers for US launch services, both predict using reusable rockets in the near future. For example, SpaceX's Falcon 9 rocket is designed to have a recoverable and reusable first stage. See bit.ly/1Jq9EjT for the historic first landing of a Falcon 9 first stage on December 21, 2015. Much of this effort is cost driven because typically about 70% of the cost of the rocket is related to the first stage. It should be noted that this is not the first attempt at reusability. The Space Shuttle (see go.nasa.gov/2r8E4aH) had reusable parts, the three main engines, which were removed between flights for extensive checking. This process was expensive and took several months, whereas SpaceX's current goal is to go from recovery to relaunch in 24 hours. However, as yet very little is known about how the launch acoustics change when reusing hardware or how the vibroacoustics might potentially be more damaging to hardware that has already been used and structurally stressed.

Finally, rockets are getting larger and louder. NASA's latest rocket, the SLS (see go.nasa.gov/2365V9K), will be the most powerful they have ever built, with 20% more thrust at liftoff than the Saturn V. That is, at liftoff, the SLS will generate more than 30 times the total thrust produced by a 747 airplane! Such extremely high fluctuating acoustic loads are a principal source of structural vibration, and this vibroacoustic interaction critically affects the correct operation of the rocket and its environs, including the vehicle components and supporting structures. Even relatively small reductions in the rocket launch noise level can result in substantial savings by reducing unexpected repairs, operating costs, and system failures. These benefits are spurring the development of novel and effective acoustic suppression methods, new experimental data-gathering techniques such as acoustic beamforming, and a plethora of mathematical modeling tools aimed at accurate noise prediction.

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BioSketch

Caroline P. Lubert is a professor of mathematics at James Madison University, Harrisonburg, VA. She obtained her degree in mathematics and her PhD in engineering from the University of Exeter, Devon, UK. Her aeroacoustics laboratory focuses on the noise emitted by turbulent Coanda wall jets, which are used in a multitude of industrial applications but, unfortunately, also often produce unwanted and harmful noise. Her research has recently focused on modeling the turbulence-generated noise emitted during lift-off of the Antares rocket, which resupplies the International Space Station and launches from Wallops Island, VA.

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Advancements in Thermophones: Sound Generation from Nanoscopic Heaters

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Researchers adapt solid-state sound-generation techniques discovered shortly after the invention of the telephone.

Introduction

It is often stated that "nothing is new" to portray the idea that everything that we do and learn is, in one way or another, a recycled version of something that someone else has done before. I find this statement to be true everywhere I look, and in the instances in which it didn't appear to be true, it was only because I hadn't looked hard enough. Of course, there is nothing wrong with this. It seems only natural that we learn through imitation before we can then adapt what we've learned to something else by "standing on the shoulders of giants," to borrow a metaphor. Even, at times, when our insights seem serendipitous or of our own accord, another has already come across the same concept before. Such was the case when, in 2008, a group from Tsinghua University in Beijing, China, passed an alternating electric current through a thin, transparent sheet of carbon nanotubes (CNTs) and discovered that it produced sound (Barras, 2008). With a bit of investigation, they found that such a device, called a thermophone, had existed for over a century but that the modern nanomaterial had simply made it much more efficient than those of previous generations. In fact, there is currently an entire field of physics called thermoacoustics that is undergoing a revitalization and progressing because of advancements in modern technologies such as lasers, computing, and very large scale integration (VLSI) that enables the manufacture and patterning of nanoscale materials. Thermoacoustic research shows promise for new devices as well as alternative takes on existing devices. Such devices range from biomedical imaging tools and sonar transmitters to engines to refrigerators.

Thermoacoustics

Thermoacoustics, or study of the interaction between heat and sound, has been a curiosity of individuals long before the "self-explanatory" term was made popular by Rott (1980). Recorded observations of sound generation by heat date as far back as 1568 when a Buddhist monk described tones generated by a ceremonial Japanese rice cooker (Noda and Ueda, 2013). A video demonstration using a handmade variant of these rice cookers is can be seen at bit.ly/2NNGCHv.

The discovery of "singing flames," which is flame heat inducing air motion along tubes or jars to produce sound, is attributed to Higgins (1802). This article reported the effect, along with a letter from Higgins claiming the initial discovery in 1777, which had become somewhat of a novelty demonstration by his students (Higgins, 1802). The article describes placing the neck of various sized jars at some distance over a hydrogen gas flame that produced "several sweet tones." Various publications surfaced attempting to explain the mechanism of sound production in terms of water vapor evaporation and condensation or a series of small com-

bustion explosions, none of which held up successfully to scrutiny. It was not until after experiments were performed by Sondhauss (1850) and Rijke (1859) that an adequate theory was developed by Lord Rayleigh (1878).

Experiments were performed by Sondhauss and Rijke between 1850 and 1860 resulting in the setups that bear their names. A Sondhauss tube is a thermoacoustic device with a long cylindrical neck that is closed off at one end, sometimes in a bulbous structure (**Figure 1A**). When the bulb or closed end is heated, sound may emit from the opening in the neck, with a frequency dependent on the resonant structure created by the bulb and neck. This effect had long been known by glassblowers who, at times, noticed a sound produced as blown glass bulbs began to cool. In the Sondhauss tube, a parcel of cool air enters the heated bulb and heats up as it is compressed further into the bulb. The heated parcel subsequently expands and further cools as it comes in contact with the colder tube. The rarefied region then collapses again toward the bulb as it heats, completing a cycle. Rayleigh posed that the condition that enables the amplification of sound (which occasionally needs a small "kick" to jump start) is that the parcel is heated during compression and cooled during rarefaction by displacing to the hot and cold regions of the tube, respectively.

A Rijke tube is a simple vertical cylinder open on both ends, with a heat source, often a heated wire mesh or gauze, inserted in the bottom half of the tube (**Figure 1B**). Here, convective flow rises through the bottom of the tube and is heated as it passes through the mesh. The gas first expands and then contracts as it interacts with the sidewalls of the tube. Eventually, a standing wave is created (which can be mathematically represented as a combination of traveling waves moving up and down the tube, being partially reflected at the openings) superimposed on the convective flow. Early embodiments of both devices by Herschel (1874) can be seen in **Figure 1C**.

The source of heat for "singing flames" was originally through the combustion of hydrogen gas, although, the ability to produce an acoustic response is largely independent of how the heat is generated. Instead, what matters most is how heat is distributed and how it propagates throughout the system. Some Rijke tubes, even early on, used battery-powered resistive heating (Joule heating) of coiled wires. Furthermore, if the heating element is fine enough, sound can be produced without the resonating tube by oscillating the temperature of the element at acoustic frequencies. Such a device is called a thermophone, a term coined by Weisendanger (1878a,b).

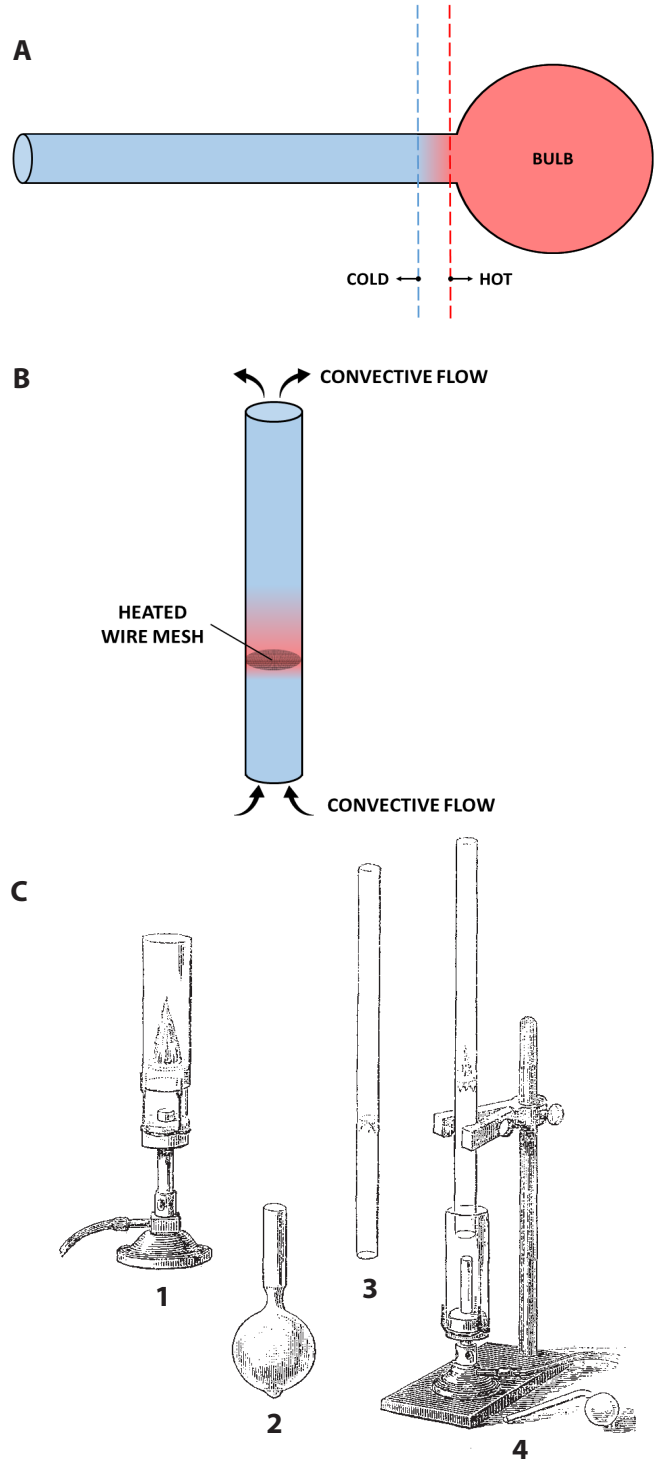

*Figure 1: **A**: a simple bulb-end Sondhauss tube. Typically, a flame is used to heat the bulb externally. **B**: a Rijke tube with open ends and a heated wire mesh inserted in the bottom half of the tube. **A** and **B**: heuristic representations of the hot (**red**) and cold (**blue**) regions and are not accurate representations of the temperature profiles. See text for details. **C**: various embodiments of Rijke and Sondhauss tubes, although not referred to as such, by Herschel (1874). **1** and **4**: Bunsen burners, emit a flammable gas that rises into a short (**1**) or long (**3** and **4**) vertical Rijke tube. The gas combusts at the hot wire gauze located partway up from the bottom of the tube. Even after the gas is shut off, a loud sound can be heard that fades as the gauze cools. **2** and **4**: Two short bulbous Sondhauss tubes.*

Thermophones operate by rapidly changing the temperature of an electrically conducting heater element, be it a wire or thin sheet, which interacts with gas in its immediate vicinity. This heated gas rarefies or expands and then cools and contracts again in accordance with the ideal gas law as the current through the heater is decreased. A video by Michigan Tech Acoustics demonstrates a simple thermophone playing music at a conference exhibition (see acousticstoday.org/mDEcx). A driving requirement for thermophones is that the heater element have a low heat capacity and a large surface area by which it can exchange energy with the surrounding gas. It is perhaps ironic that Weisendanger's (1878a,b) original thermophone, which was incited by the excitement surrounding acoustics following Alexander Graham Bell's communication on the telephone two years before, was claimed to operate due to a thermally induced dimensional change in the wire itself, modernly characterized by a materials coefficient of thermal expansion.

Although it is possible that Weisendanger's (1878a,b) thermophone was enhanced at certain frequencies due to a dimensional change in the wire, modern thermophones don't rely on mechanical actuation of the element itself. There was, in fact, much confusion and doubt as to what the particular transduction mechanism was at the time. Preece (1880) reported that it was noted by De la Rive, in 1843, that sounds were produced by passing current through iron wires, but the effect was attributed to magnetism. Preece (1880) also reported that Bell suggested straight pieces of iron, steel, and graphite could also produce sound when driven by a battery.

Bell and Tainter (1880) also presented the photophone in 1880, a device in which intermittent light, that is, light modulated by a chopper or fan, impinging on a thin disk of nearly any hard substance would produce a sound of frequency corresponding to the modulation rate. Bell considered this one of his greatest inventions. Even by 1898, Braun (1898), to whom many have presumptively attributed the invention of the thermophone, described the acoustic sound as being partly produced by a change in length of wire. It becomes understandable then, especially considering the limitations of observing thermal changes at acoustic frequencies, that it was uncertain as to which mechanism produced sound, temperature fluctuations causing mechanical strain in the material or temperature fluctuations causing mechanical strain in the air.

In actuality, both mechanisms mentioned above occur to one degree or another. Which of these a user wishes to interrogate is the subject of various photoacoustic techniques. For example, in thermal wave imaging, a sample containing optical absorbers is placed in a water-filled cavity with ultrasound detectors placed along its edge. Short laser pulses excite the sample-producing acoustic waves due to thermoelastic expansion of the material that is recorded by the detectors. In photothermal beam deflection spectroscopy, the refractive index gradients in a coupling liquid produced by the "mirage effect" will deflect a laser beam that is near the sample surface. In a gas-microphone approach originating from Bell, periodic or intermittent monochromatic light impinges on a sample, is absorbed, and thus produces periodic heating. Heat diffusion to an adjacent inert gas then produces thermal rarefactions and compressions in the gas as a thermally driven acoustic wave. The acoustic signature is recorded using microphones mounted flush within a resonant absorption cell that houses the sample and inert gas.

These various photoacoustic techniques can be utilized for imaging, spectroscopy, or material characterization. Initial photoacoustic theory established in a series of articles by Preece (1881) and Mercadier (1881a-c) was more comprehensively formulated many years later by Rosencwaig and Gersho (1976). This has led to various applications for photoacoustics, particularly in regard to biomedical imaging applications. Manohar and Razansky (2016) provide a much more extensive historical review of photoacoustics for the interested reader.

The Thermophone

The first quantitative theory for thermophone sound production was developed by Arnold and Crandall (1917), which paved the way for the thermophone to be used as a functional device. Since then, the thermophone has historically found most use as a precision source of sound for microphone calibration. Such thermophones consist of an active element, such as gold leaf or thin platinum wires, suspended above a metal backplate that is then coupled with a front plate housing the microphone element to be calibrated. Two narrow capillaries in the backplate serve as an inlet and outlet to supply hydrogen gas to the cavity formed when the two sides of the device are brought together. Hydrogen gas has a much higher sound speed than air and shifts the internal cavity resonances higher in frequency, thereby extending the usable bandwidth of the calibration instrument. One of these thermophones on its backplate is shown in **Figure 2A** along with a diagram in **Figure 2B**.

The usefulness of a thermophone is due to its predictable and relatively smooth frequency response over a wide band-

width. This is primarily the result of a lack of mechanical moving parts that always have accompanying passive structural resonances. Normally, these resonances are "pushed out" of the band of interest through a combination of material choice and appropriate sizing of structural parts. For thermophones used as a precision source of sound, the only dimensions of concern that would limit bandwidth are those of the cavity encasing the thermophone and microphone element being calibrated. It remains today that one of the most attractive features of thermophones is that their acoustic response is largely decoupled from any mechanical parts.

Thermophones saw utility as a precision source of sound but were never widely used for any other purpose due to their poor efficiency compared with the electrodynamic loudspeaker and other more conventional transduction sound sources. As with some other transducers, thermophones are also hindered when it comes to reproducing arbitrary waveforms due to their intrinsically nonlinear transduction. The acoustic response is quadratic with respect to the driving voltage or current and requires a DC bias to linearize the response. Similar to what occurs in a variable reluctance transducer, this bias current continuously generates excess heat and reduces efficiency. Other modern conventional signal-processing techniques such as amplitude modulation, pulse width modulation, and pulse amplitude modulation can be used as well to rectify arbitrary signals. Bouman et al. (2016) provide a comparison of their thermophone's efficiency with a biased input signal versus an amplitude-modulated one. Even with rectified signals, however, the various frequency components of an arbitrary input signal will still generate corresponding heterodynes as well as second harmonics. Although typically undesired, such nonlinearities can be exploited to probe material properties (Heath and Horsell, 2017).

Modern Thermophones

Relatively few articles concerning thermophones were published after the 1940s until Shinoda et al. (1999), inspired by photoacoustic studies on porous silicon, presented a porous doped silicon thermophone for ultrasonic emission. Interest grew substantially after Xiao et al. (2008) from Tsinghua University reported a flexible thermophone with an active element composed of a CNT sheet. CNTs are nanoscopic or nanoscale cylinders of carbon atoms arranged in a hexagonal lattice (much like a piece of wrapped chicken wire) that can have exceedingly high aspect ratios.

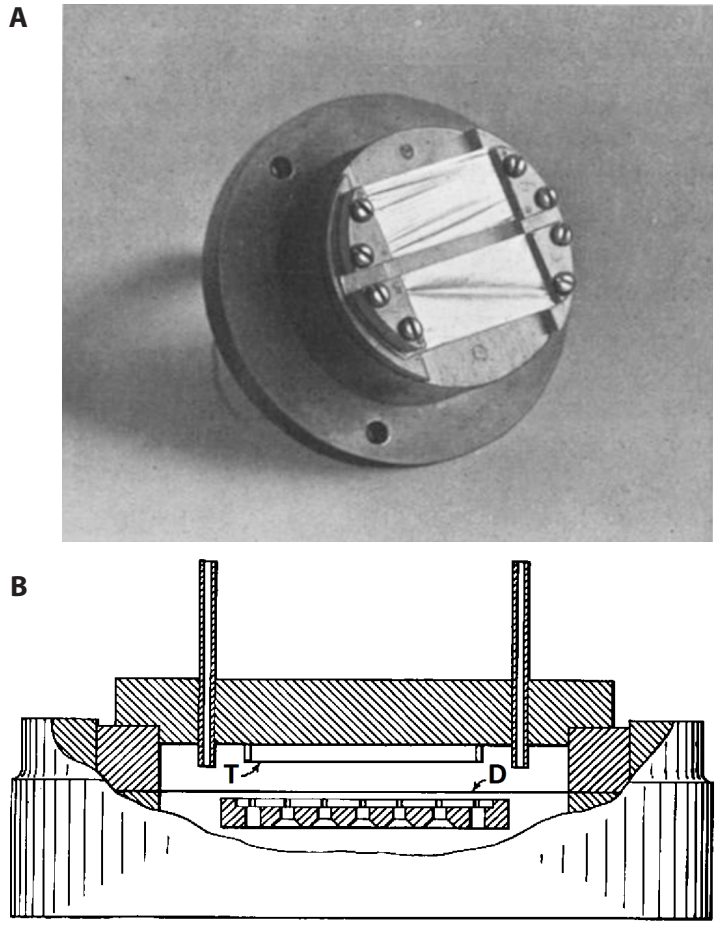

Figure 2. ***A****: thermophone on its backplate used as a precision source of sound for microphone calibration (Ballantine, 1932).* ***B****: a schematic of an assembled thermophone and microphone from Sivian (1931). T, thermophone active element; D, electrostatic microphone diaphragm. Images reprinted from Nokia Bell Laboratories, with permission.*

For example, CNTs used in thermophones have a diameter on the order of 10 nm (about 1/10,000 the thickness of a human hair) but are hundreds of microns in length. A CNT "sheet" useful for thermophone applications is made from a CNT "forest" (a highly oriented dense vertical array of CNTs) made by a process called chemical vapor deposition (CVD). The CVD process for growing the CNTs now used in thermophones was adapted by Zhang et al. (2005). Catalyst nanoparticles are deposited on a silicon wafer and individual CNTs grow vertically from these catalyst particles during the CVD process as a heated feedstock gas is passed over the wafer. With careful adjustment of various growth parameters, the edge of the resulting forest can be mechanically pulled into a less dense porous sheet of horizontally aligned CNTs (**Figure 3**). The roughly 50-μm-thick CNT sheet can be con-

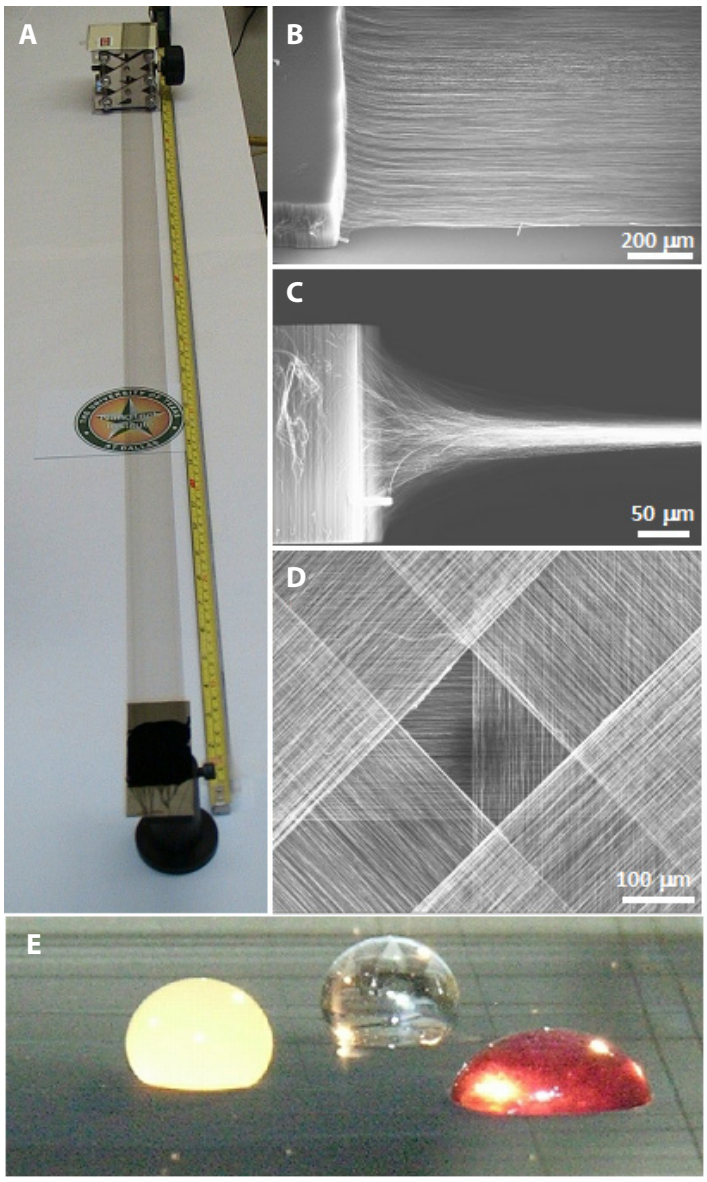

Figure 3. ***A****: 1–meter-long freestanding carbon nanotube (CNT) sheet pulled from the edge of a CNT forest by Zhang et al. (2005).* ***B*** *and* ***C****: scanning electron microscope images of the interface between the horizontally aligned sheet and the vertically aligned forest, respectively.* ***D****: layering or stacking of CNT sheets at various angles.* ***E****: hydrophobic CNT sheet can support droplets from various aqueous solutions. Images reprinted from the AAAS, with permission.*

tinuously drawn from the forest and, if desired, spun into fibers. Individual CNTs can be semiconducting or metallic, depending on their chirality (i.e., the relative orientation of the 2-D lattice to the angle in which the lattice is "wrapped" on itself), but, statistically speaking, a random array of various chirality CNTs is electrically conducting (Saito et al., 1998). The high porosity of CNT sheets affords them a much larger interface with the surrounding gas and results in more efficient acoustic thermophones than those that utilized thin platinum wires or gold leaf, although this efficiency is still well below that of most conventional transducers.

Although CNT sheets are considered to be very mechanically robust by certain metrics (and can support droplets of liquid 50,000 times the weight of the sheet itself), their extreme porosity leaves them vulnerable to damage by even small macroscopic natural events such as a droplet of water falling on the sheet or a moderate gust of air. In an attempt to remedy this problem, other thermophone heaters that have been manufactured and examined include graphene sponges (Fei et al., 2015), CNT sponges (Aliev et al., 2015), carbonized electrospun polymers (Aliev et al., 2016), and carbon fiber (Dzikowicz et al., 2017). These denser but also more manageable materials highlight the tradeoff between mechanical robustness and thermoacoustic efficiency.

Another parameter thermophone designers must take into consideration is device scalability, which asks how quickly and efficiently these materials and devices can be manufactured and assembled in a repeatable fashion. Since Shinoda et al. (1999), various other thermophone active elements have been produced using VLSI technology such as multilayer (Tian et al., 2011a) and single-layer (Suk et al., 2012) graphene sheets, tungsten thin films deposited by atomic layer deposition (Brown et al., 2016), and thin gold (Dutta et al., 2014), silver (Tian et al., 2011b), and aluminum (Niskanen et al., 2009) wires.

The group from Tsinghua University who published the first CNT-based thermophone element described, in a patent submitted shortly thereafter, sound produced from such a device when submerged just beneath the surface of water (Jiang et al., 2008). Aliev et al. (2010) drew the conclusion that such an effect is possible underwater because carbon nanotubes are hydrophobic and sustain a thin layer of air that then thermally expands on heating as opposed to relying on the thermal expansion of water, which is quite negligible below vaporization. Although sound can be discerned from a pristine CNT thermophone that has been submerged just below the surface of the water, removing the CNT sheet from the water results in physical damage to the element.

An alternative to producing more robust thermophone elements is to shield the fragile element from the external environment. To protect CNT sheets, Aliev et al. (2010, 2014b) and Mayo (2015) have encapsulated them between various materials ranging from polyimide film and mica sheets to ceramic and metal plates. Introducing the encapsulation media results in a mechanical system with resonances that

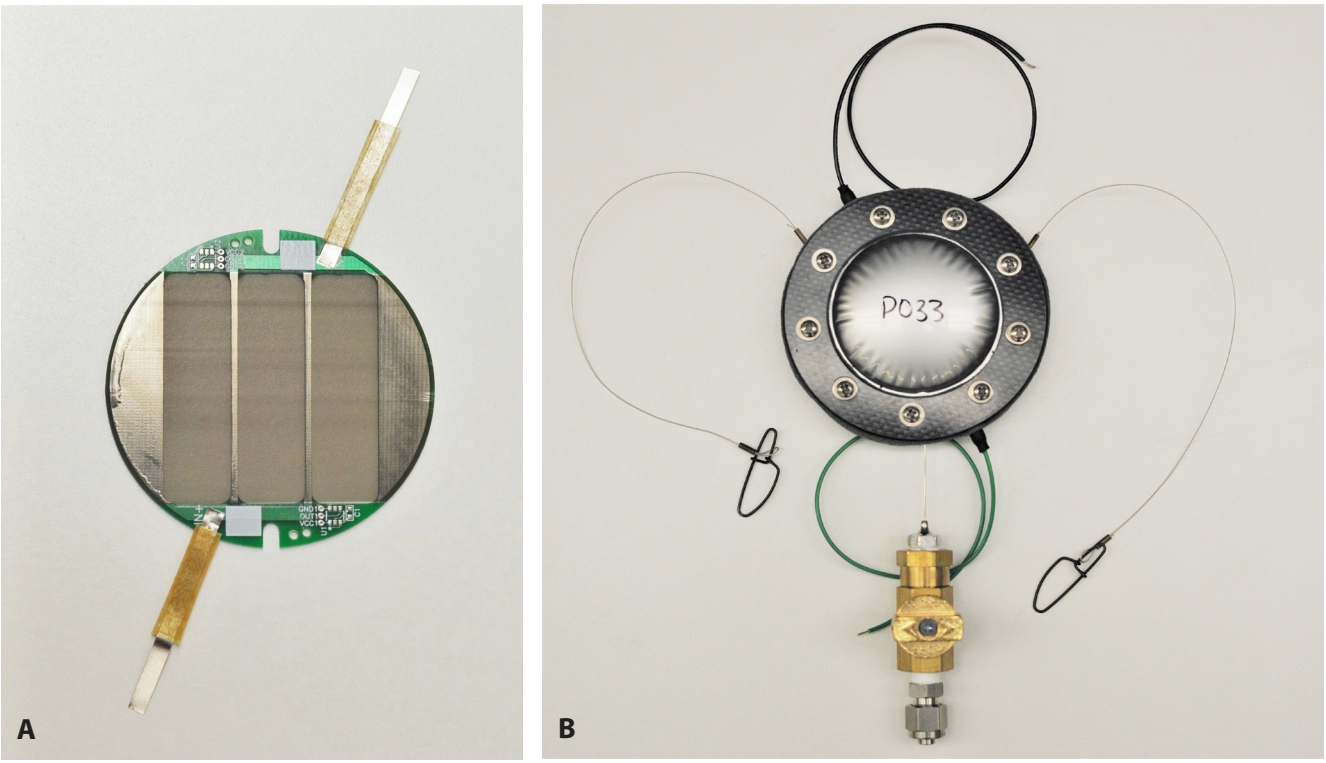

Figure 4. ***A****: 6.35-cm-diameter G10 circuit board substrate populated with CNT sheets. The device resistance is specified by the separation gap and height along with the number of overlapped CNT sheets.* ***B****: fully assembled aluminum laminate thermophone pouch (labeled P033) inflated with argon gas and supported along its perimeter by a black carbon fiber ring. The* ***black*** *and* ***green*** *leads have been soldered to the positive and negative tabs, respectively. The needle port along with the brass valve below the pouch allows the pouch to be pressurized with inert gas. The two metal wires with clips loosely connect the thermophone to a rigging apparatus (not shown) for calibrated acoustic testing.*

can be tuned by choice of the properties and dimensions of the encapsulation media. The advantage of the thermoacoustic approach is that these mechanical resonances are independent of the properties of the active material.

Aliev's encapsulated thermophones drew interest from the US Navy, who have worked to evaluate and adapt the technology for use in underwater sonar applications. Recent free-field acoustic calibration presentations have looked at the usefulness of various aluminum laminate thermophones as underwater projectors (Howarth et al., 2016). One such adaptation (**Figure 4**) is something of a cross between the thermophones used for microphone calibration (**Figure 2**) and bubble transducers demonstrated by Sims (1960). These thermophones consist of a CNT-active element suspended across a substrate (**Figure 4A**) that is then housed inside an aluminum laminate composite and pressurized in an inert gas environment (**Figure 4B**). These devices act as low-frequency resonating-bubble projectors that allow them to achieve relatively large source levels for their small package size (143 dB re 1 μPa at 1.4 kHz from a single 6.35-cm-diameter thermophone). When the inert gas within the pouch is replaced with a liquid, the bubble is removed and the device becomes a broadband projector (Mayo et al., 2017).

The main selling point of thermophones is just that, their cost. The simplicity of thermophone design (you could literally make one with a fine wire and a power source) and the small amount of active material required puts a very low cost floor on production. Thus, thermophones can be made very thin and lightweight, use no rare earth metals, and can easily conform to most surfaces. In contrast, although piezoelectric ceramics dominate the underwater projector market due to their high electroacoustic conversion efficiencies, most use lead or other heavy metals and require complex processing steps to manufacture high-quality material such as single crystal ceramics.

Theory

Arnold and Crandall (1917) along with Wente (1922) were the first to significantly develop a theory for thermophone transduction as a precision source of sound. Their calcula-

tions predict an acoustic pressure that is proportional to the square root of the driving frequency (of a biased system). More recent models of CNT thermophones in free space were published by Xiao et al. (2008), Vesterinen et al. (2010), and Aliev et al. (2013). Each of these models differ slightly, but under the same basic assumption of a negligible active-element heat capacity, the far-field acoustic pressure is linearly proportional to its frequency and reduces to

$$p = \frac{f\, P_{el}}{2\sqrt{2}\, r\, c_p\, T_{amb}} \qquad (1)$$

where p is the acoustic pressure at a distance r from the monopole source that is driven with an electrical input power P_{el} producing an acoustic signal at frequency f in an ambient gaseous environment at temperature T_{amb} with gas-specific heat capacity at constant pressure c_p. Therefore, in the "absence" of the thermophone active element, the thermoacoustic transduction process is only determined by the drive power, frequency, and properties of the surrounding gas.

These formulations are limited in scope to sources small with respect to the acoustic wavelength and the ability of the thermophone to maintain its background ambient temperature. At high power, the acoustic pressure will thermally saturate as the background temperature approaches the surface temperature on the heater, eventually causing the active element to degrade, either burning or melting in extreme cases (Aliev et al., 2014a). Both Xiao et al. (2008) and Aliev et al. (2015) have proposed that the difference in the frequency dependence between various thermophones is due to a more substantial heat accumulation of the active elements used in the early 20th century compared with the CNTs often utilized in thermophones today.

Models for the far-field acoustic pressure of encapsulated thermophones are studied less and, although few exist, none appear robust enough to predict the performance of devices that vary significantly in dimension or housing composition. Even fewer models of underwater thermophone projectors exist, an area much in need of development for any practical implementation of thermophones in naval applications.

The largest criticism of thermophones by far is their low efficiency. People often think, "can you just make a better nanomaterial that converts heat to sound more efficiently," but often it isn't the element itself that is the problem. At the current stage of development, the efficiency of a thermophone open to its environment (i.e., not encapsulated) appears to be limited by the properties of air rather than the active element.

To complicate things further, it is difficult to provide a comparison of thermophone performance to that of conventional transducers. Most transducers operate within a region of essentially constant efficiency no matter what power is provided to the device. Thermoacoustic devices are completely different, and the conversion process is similar to that of a car engine or power plant that has a behavior limited by Carnot's cycle in which the efficiency is dependent on the temperature difference between the "hot" fluid and "cold" reservoir. Therefore, so long as the background temperature of the "cold" reservoir surrounding a thermophone is maintained, an increase in input power provides a proportional increase in efficiency. For acousticians, this translates to a 6 dB increase in sound pressure level (SPL) for each doubling of input power as opposed to the 3 dB increase seen in conventional devices.

As discussed, thermophone efficiency may be greatly increased by operating at a higher frequency and higher power and by creating resonant devices. However, efficiency still remains orders of magnitude lower than in conventional devices, and each of these requirements limits application potential. It is for this reason that commercialization of thermophones at this point has been stymied.

Conclusions

Thermophone transducers generate acoustic signals by modulating the temperature of an active element via Joule heating. Heat transfer to gas adjacent to the element causes thermal rarefactions and compressions producing an acoustic wave. The historical origins of these transducers are entangled with the invention of Bell's telephone and scientific observations that followed, leading to the development of photoacoustic spectroscopy, thermoacoustic engines, and thermoacoustic refrigeration. Arnold and Crandall (1917) developed the theoretical foundation for sound projection by thermophones that enabled their use as a precision source of sound for microphone calibration. Decades went by with relatively few developments in thermophone technology until highly porous nanoscopic materials such as porous doped silicon and carbon nanotubes were utilized in the late 1990s and 2000s, respectively. The discovery that such materials significantly improve thermophone efficiency has led to a resurgence in interest as well as new theoretical models and potential use cases.

Overall, low efficiency, the mechanical fragility of highly porous thermophone heaters, and an effective lack of receiving capability has limited thermophones from making their way into any practical commercial devices. Still, a few niche use cases exist in which thermophones could outshine their traditional counterpart projectors due to their broadband response and low manufacturing cost. Use as an underwater sound projector place thermophones in an ideal environment where they can be run in their most efficient regimen at high power with ample cooling capability.

Moreover, most thermophone technologies are easily upscaled with various active heating elements being produced using VLSI processes. New materials are being explored as more mechanically robust thermophone elements, although freestanding CNT sheets currently remain the most efficient transduction material. Thermophone encapsulation provides a means of protecting the relatively fragile active material from harsh environments but also results in a resonant device. This resonance can be tuned independently of the active material that is usually suspended from a substrate. Thermophone elements are usually arrays of wires or planar films that are suitable for making large area projectors that are very thin and lightweight.

The future of thermophone projectors is still largely unknown. The ability to generate sound without any mechanically moving parts makes thermophones a fascinating technology to study for potential applications. However, modern thermophones are still a relatively new technology and are certainly not an end-all replacement for conventional devices. Indeed, an inspection of recent thermophone publications shows that most studies on the topic have been conducted from a physics or materials science perspective and not for direct applications. Thus, additional evaluation and critique by trained acousticians and engineers is sought to more rigorously quantify thermophone performance and help progress this exciting technology. Along with developing the theoretical foundation of thermophones, input from biologists, sonar technicians, medical doctors, and many others is needed to highlight the various niche areas in which the advantages of these projectors can be utilized. Only time will tell as to what other practical devices this technology can produce. In the meantime, it continues to provide a very curious tabletop demonstration for students.

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BioSketch

Nathanael Mayo is a research scientist and experimental physicist at the Naval Undersea Warfare Center (NUWC) in Newport, RI. Dr. Mayo studied physics at the University of Texas at Dallas where he worked at the Nanotech Institute and earned his doctorate focused on thermoacoustic sound generation using conductive nanomaterials (thermophones). Since graduating, he has worked within the Devices, Sensors, and Materials R&D branch at the NUWC, developing and testing carbon nanotube-based thermophones for underwater applications. His other interests include conventional transducer design, textured ceramics, signal processing, and surface chemistry.

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Meet Sandra Gordon-Salant

In this edition of "Ask an Acoustician," we hear from Sandra Gordon-Salant. Sandy is a professor in the Department of Hearing and Speech Sciences at the University of Maryland, College Park (hesp.umd.edu). She is well known for her studies on speech perception and aging. Sandy is a Fellow of the Acoustical Society of America (ASA) and a Fellow and Honoree of the American Speech-Language-Hearing Association (ASHA) and has won numerous awards for her research and teaching activities. Sandy can tell you the rest.

A Conversation with Sandra Gordon-Salant, in Her Words

Tell us about your work.

My position as professor of Hearing and Speech Sciences at the University of Maryland encompasses the three prongs of academia: research, teaching, and service (administration), although the majority of my time is devoted to research. My research program broadly examines factors that contribute to speech perception difficulties of older people in everyday listening situations (Gordon-Salant and Fitzgibbons, 1993, 2001). The underlying theory driving much of this work is that auditory temporal-processing deficits of older people contribute to altered perception of speech (Gordon-Salant et al., 2008). Work in the lab also examines the impact of reduced audibility associated with age-related hearing loss and age-related cognitive decline in specific cognitive domains to understanding fast speech, accented speech, reverberant speech, auditory-visual asynchronous speech, and speech in a background of competing talkers (e.g., Gordon-Salant et al., 2010, 2017). References and links to all of my published work can be found on my lab website at umdhearinglab.com. Also, see the article in this issue of *Acoustics Today* by Anderson, Gordon-Salant, and Dubno that talks about some of our work.

Describe your career path.

I grew up in an era in which the double standard was firmly entrenched in our society and in my parents' home in Plainview, NY. There were three professions that women could pursue: teacher, nurse, or secretary. My parents decided I would become a speech-language pathologist in the public schools, which in their view was the best teaching job possible. I was not the rebellious type and followed my parents' plans for me. As an undergraduate at the University at Albany, State University of New York (SUNY), however, I was completely mesmerized by my courses in audiology and hearing science and was decidedly unenthusiastic with the curriculum in speech-language pathology. Following my new-found passion, I obtained a master's degree in audiology at Northwestern University (NU), Evanston, IL, and completed a one-year clinical fellowship at Gallaudet University, Washington, DC. During that year, I had a chance encounter at an ASHA convention with one

of my former professors at NU, Tom Tillman, that changed my path. He convinced me to return to NU for a PhD and I jumped at the chance. I always thought I would obtain a PhD, but this single discussion with Dr. Tillman was all the encouragement I needed to pursue my further education straightaway. On returning to NU, I chose to work with Fred Wightman, who taught me how to be an experimentalist and an independent thinker. I became good friends with the other students working in Fred's lab, especially Pete Fitzgibbons and Larry Humes. When I was ready to defend my dissertation, Pete informed me of a faculty position at the University of Maryland. I got the job, which has been my one and only academic position for my entire career.

What is a typical day for you?

This is not an easy question to answer because every day is different and varies between the academic year versus summer or winter sessions. Nonetheless, many days have commonalities. I start each day checking email (who doesn't!) to learn what calamities may be awaiting me in the lab (usually first thing Monday morning), what tasks someone wants me to do, etc. I often spend time in the lab to troubleshoot problems and develop new experimental protocols. Typically, I have advising appointments with students, ranging from undergraduates considering graduate school options to PhD students planning their dissertation research. It is a rare day when I do not meet with at least one student. Much of my time each day is spent designing studies, reviewing pilot data, analyzing final datasets, and writing or revising manuscripts. These research-related activities are often in consultation with students and colleagues and are the best part of any day. I tackle at least one administrative task each day. As director of the Doctoral Program in Clinical Audiology, I monitor student progress, the quality of the instructors in the program, course scheduling, and the graduate curriculum and respond to emerging issues as they arise.

As codirector of a T-32 Institutional Training Grant (ccebh.umd.edu), I may plan a workshop, prepare an annual report, or monitor the budget. During the academic year, I prepare for teaching my class one or two days/week to keep the material current and engage the students through new techniques. I also attend at least one meeting a day, which, like for most of us, is the least favorite part of my day. These can range from department-wide faculty meetings to administrative meetings at the college or university level. I serve on promotion and tenure-review committees, curriculum committees, and award selection committees. At home in the evening, I often write reference letters for students or colleagues, review manuscripts, and review grant applications. Each and every day goes by incredibly fast.

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

The results of our experiments often do not yield the results we expect. I am no longer surprised or disappointed by an unpredictable outcome because it is inherent in conducting empirical research. Our strategy is to design experiments in a way that the results tell us something important and new, even if the outcome is unexpected. We often are prepared for the possibility of unanticipated findings as a result of collecting and monitoring a considerable amount of pilot data, and we may alter the experimental protocol as a result of these pilot data, such as adding measures to help us explain the data.

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

Work-life balance is difficult to achieve, especially for women who are early in their academic careers while at the same time raising young children. When my children, Brian and Maida, were young, my strategy was to focus on work when I was at work and focus on my family when I came home. I prioritized important events in my children's lives. I also had a lot of support from my husband, Steve, but as his own career as attorney, magistrate, and judge became more demanding, the time he could devote to family life was considerably reduced. But in looking back, I realize it was extremely stressful to have a productive academic career while at the same time enjoying a rich family life. There really aren't easy answers, but one approach is to say "no" to requests to serve on committees or write manuscripts that are not valued in academia. Now that my children are grown and living independently, I have a new work-life balance, namely, taking advantages of opportunities to enjoy my personal life while working on an increasing number of research activities. I still feel guilty saying no to work requests, but I am finally learning how to protect my time after decades in academia.

What makes you a good acoustician?

I think a lot about the acoustic characteristics of the signals we present to participants in our studies and how the individual's auditory capabilities will enable them to process these signals. We prepare many new speech materials in my lab. I insist on equalizing the levels of the speech signals within a stimulus set and characterizing the temporal properties of these signals. My students and I listen to the speech materials we develop to ensure that they are free of distortion and accurately represent the intended signal. We also collect

pilot data from naive listeners to verify speech intelligibility, list equivalence, and perceived degree of accent. Finally, we calibrate our signals daily in the lab before presenting them to listeners. Knowing the acoustic attributes of signals, signal presentation levels, and auditory capacity of listeners enables me to have confidence in the experimental results.

How do you handle rejection?

I say a few expletives and set aside the rejection letter/review for at least a few days. Then I start to consider the reasons specified for the rejection and whether they can be addressed in a revision either as a manuscript sent to another journal or as a resubmitted grant application. Inevitably, the comments stated in the rejection letter are quite helpful in revamping the submission, and, as a result, I have often been successful in overhauling the submission.

What are you proudest of in your career?

I am proudest of three aspects of my career: the impact of my research on improving our understanding of age-related hearing loss, my influence on students, and my success in obtaining external funding to support my work. My entire research career has been focused on elucidating the mechanisms underlying age-related hearing loss (as mentioned in ***Tell us about your work***), and I believe that our theories about age-related decline in auditory temporal processing have now become broadly accepted. It is quite remarkable to see my work referenced in articles that advance the scientific premise I espoused, and I am also gratified to learn that some of my work has impacted audiology practice. My students are a great source of pride for me. I have watched some of them develop from eager but uninformed undergraduates to knowledgeable and insightful PhD candidates with publications of their own. I am quite honored to have worked with so many students and to have helped launch their research careers. My success in obtaining external grant support is frankly beyond my imagination. Perseverance and unusual ideas, coupled with collegial support, have helped me achieve continuous funding for nearly my entire academic career. I am humbled and proud of this record.

What advice do you have for budding acousticians?

My first piece of advice is to follow your passion. If you are excited about a particular area of research, then channel your energy into that research focus. Excitement about your research program is evident in talks, manuscripts, and grants and will motivate you to think deeply about your research questions.

My second piece of advice is to aim high. Budding acousticians should submit manuscripts to the premiere journals in the discipline such as *The Journal of the Acoustical Society of America*. Gaining acceptance in these journals requires more work than acceptance in other journals but is well worth the effort in bolstering your reputation. You should start early and often to submit grant applications to federal agencies. Expect rejection and grow a "thick skin," meaning, don't take the rejection personally but learn from it and submit a better application next time. My observation is that young investigators who consistently write grant applications ultimately get funded. You just need to persevere! Third, volunteer to serve as a reviewer for top peer-review journals and federal grant agencies. These are invaluable learning experiences for writing a better manuscript or preparing a better grant. Finally, say "no" to requests of your time that don't advance your career or that interfere with your work-life balance. How do you say no? My advice is to reply that you will think about the request and will give a response within a week. This provides an opportunity to ponder whether you really want to do what is asked of you and whether it is a task that will advance your career. It's also easier to give a definitive "no" after some time has passed.

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

Most of us think in terms of a 5-year plan; mine includes several objectives. I hope to accomplish continued success in carrying out the work outlined in my grants and contracts and in submitting one grant renewal. As part of this objective, I would like to see a number of manuscripts published describing work that has been conducted in my lab over the years but hasn't yet been published. I aim to work with my current group of PhD students toward completion of their degrees. I am committed to helping them begin successful careers in research. Finally, I would like to inspire junior colleagues toward a successful career in academia by advising them on grant applications and consulting with them on career choices.

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ASA ACOUSTICAL SOCIETY OF AMERICA

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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.

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Acoustical Society of America Chapters: Form and Function

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History

When the 416 charter members of the Acoustical Society of America (ASA) first organized, their goal was "to increase and diffuse the knowledge of acoustics and promote its practical applications" (Schmid and Moran, 2006, p. 36). Few could have predicted the role the Society and its members or their scope of research would play over the next 89 years.

After World War II, with membership and research in acoustics at an all-time high, the ASA formed the "Regional Sections" Committee to address the growing interest of members to form local chapters. Nearly a decade later, in 1954, the Society would enjoy three major moments in their history. The celebration of their 25th anniversary was marked by a banquet in May that was attended by 527 members, a record attendance for an event at that time (Firestone, 1954). During this same banquet, ASA founder Wallace Waterfall was named as the first recipient of the Gold Medal Award. To cap off an already exciting year, the Society saw a dramatic organizational change with the establishment of its first four local chapters.

What Are Chapters?

There are currently two types of chapters associated with ASA, regional and student. These are included as "Administrative Committees" of the ASA. The current chapters are listed in **Table 1**.

Table 1. *List of Chapters*

Regional Chapters	Student Chapters
Cascadia Chapter	Austin Texas Student Chapter
Chicago Chapter	Brigham Young University Student Chapter
Greater Boston Chapter	Columbia College Chicago Student Chapter
Los Angeles Chapter	Georgia Tech Student Chapter
Music City Chapter	Northeastern University Chapter
Upper Midwest Chapter	Ohio State Student Chapter
Florida Chapter	Penn State Student Chapter
Narragansett Chapter	Purdue University Student Chapter
Philadelphia Regional Chapter	Rensselaer Polytechnic Institute
Washington, DC Chapter	Seattle Student Chapter
North Carolina Chapter	University of Cincinnati Student Chapter
North Texas Chapter	University of Hartford Student Chapter
	University of Kansas Student Chapter
	University of Nebraska Student Chapter

For more information on and links to the individual Acoustical Society of America regional and student chapters, go to asacpters.org/asa-chapter-locations.

Regional chapters were formed to meet the burgeoning needs of ASA members and are primarily centered on supporting and representing practitioners in the acoustical community. Although some chapters have a primary focus, such as underwater or oceanography, each chapter represents a unique cross-sectional demographic of the 13 technical committees, with many chapters having a wide interdisciplinary makeup of technical committee members.

Student chapters are a somewhat recent addition to the Society, with the first one being formed in 2000. Although fairly new, student chapter members make up 13% of the ASA membership and are vital to promoting Society awareness among up-and-coming graduates as well as supporting them socially, academically, and with their professional development goals. Student chapters "promote a robust sense of community for students within the ASA" (Flynn and Young, 2018, p. 65) through numerous networking and educational events. A great example of this is the four special sessions held each year. These four events don't just introduce student members to the 13 technical committees but they also serve as an avenue into the acoustical community, with presentations by senior ASA members, poster summaries for various acoustic research programs, and even chances to secure grant/fellowship funding.

Because national meetings occur only twice a year, regional and student chapters are also a great way for members to stay involved with the Society and the acoustician community at the local level. The required minimum of three meetings per year allow regional chapters to actively engage members while keeping the momentum from the most recent national meeting going. Chapters also provide an avenue to include professions, organizations, and experts of similar interest. For instance, in Architectural Acoustics, local chapters have engaged members of both the Audio Engineering Society (AES) and Architectural Institute of America (AIA). The flexibility to engage outside organizations allows chapters and members to further engage manufacturers, consultants, and laboratories. In an effort to help promote involvement and awareness in the Society, participation in local and regional meetings does not require ASA membership.

What Does a Chapter Meeting Look Like?

There are no exact or structured guidelines for chapter meetings. This allows each chapter to develop its own identity and character best suited to serve the local membership. The following is a partial list of what you might expect or experience at your local chapter meeting.

Educational Outreach

Many chapters participate in outreach programs at the local primary and high school levels. Organizations like the Boy Scouts, Girl Scouts, YMCA, and YWCA also serve as great opportunities to reach out to the potential young engineers and acousticians of tomorrow. These programs usually include engaging, hands-on demonstrations on the physics of acoustics. A great illustration of this is the paper plate loudspeaker demonstration recently given by the Pennsylvania State University (University Park) chapter to a local Girl Scout troop as well as several local schools.

Professional Outreach

Chapters facilitate the transfer and sharing of knowledge between acousticians and professionals in related industries and fields of study. Collaborating with organizations like architectural firms, the AES, the AIA, the National Oceanic and Atmospheric Administration (NOAA), the National Recreation and Park Association (NRPA), and many more provide great teaching and learning opportunities for members.

Tours

Many chapters coordinate tours of local acoustical interest for members. For example, the Philadelphia (PA) chapter has toured the Kimmel Center for the Performing Arts and Community Loud Speaker and is considering a trip to Martin Guitar in Allentown. The Boston (MA) chapter has made several trips to Boston Symphony Hall, and the St. Louis (MO) chapter recently visited the Blues Museum. Tours can be as simple as an unguided walkthrough or as detailed as a guided or even behind the scenes tour and provide a great opportunity for our members to experience and explore the fields of acoustics outside a classroom or laboratory environment.

Technical Talks, Guest Speakers, and Lectures

We, the acoustic community, are fortunate to have numerous opportunities for the interactive synergy and sharing of ideas, learnings, and experiences. Special guests are often invited to local chapters to present as experts in a specific topic. Two great examples are when Dr. Kenneth P. Roy came to the local Philadelphia chapter to give a presentation on design strategies for architectural acoustics and when Dr. Dan Brown recently held a colloquium on underwater imaging systems at the Pennsylvania State University chapter.

Presentations are not limited to acousticians. Anyone in a related field, from educators to industry leaders, is welcome to participate and present. Regional and student chapters further provide local members an opportunity to increase awareness about the Society, share their work, learn from each other, and refine presentations that may be given at the upcoming national meeting. Some chapters even schedule a weekend-long symposium to further help members practice and refine their presentations.

Technical Projects

Popular among student chapters, members participate in a specific project or competition. This may include measurements in an existing field of study or innovating for a unique problem.

The Royster Competition

Thanks to the generous support of Larry and Julia Royster, a chapter-organized poster competition is run once a year, with $5,000 in scholarship awards. It is open to full-time graduate students enrolled in a program involving acoustics or senior undergraduate students expecting to enroll in such a program. Posters must relate to hearing conservation or noise control to be eligible, with topics including education, sound surveys, engineering and administrative controls, hearing protection, and audiometric evaluations. For more information, please visit acousticstoday.org/roystercomp.

Networking, Social and Fun

Of course, fun is part of any ASA event! Events like the Philadelphia chapter's "Jazz at the Bistro," with good music, food, and friends, give members the opportunity to socialize and get to know their colleagues. Student chapters also hold numerous events throughout the school year, such as the Georgia Tech (Atlanta) Student Chapter Spring Semester-End Seminar and BBQ or the Pennsylvania State University MATLAB Bootcamp.

Each chapter can form fit to the requirements and abilities of their members. With some chapters meeting each month and most quarterly, members gain exposure to numerous unique events, tours, and guest speakers. The sound mixture of business and social networking events ensure each participant has the opportunity to share and grow professionally and as members of the ASA.

Those wishing to start their own regional or student chapter should contact Sandra Guzman (Guzman_Sandy@shure.com) or Ken Good (kwgoodjr@armstrong.com) or check the following website for details on the application, charter, and approval process (acousticstoday.org/asachapters).

Chapters and the Acoustical Society of America Vision

Regional and student chapters are one of the most powerful opportunities to advance the ASA's Strategic Leadership for the Future Plan. Chapters meet all four of the goals identified at the Strategic Leadership for the Future Summit.

Awareness of Acoustics

Chapters bring the science of acoustics home to the local level and provide an opportunity to engage businesses, organizations, and colleagues of similar interest.

Member Engagement

Through the numerous meetings and events, members can remain engaged locally throughout the year even if traveling to a national meeting is not possible.

Dissemination of Information and Knowledge

Guest speakers, tours, and miniconferences further increase the opportunity of members to share and learn about the science of acoustics and related fields.

Financial Stewardship

Good stewardship is often mistaken for money not spent, but true stewardship is money WELL spent. Investing in the future of the acoustics community through student chapters and engaging sister organizations further promotes meeting each of the strategic initiatives.

The local regional chapters provide excellent opportunities for people with a wide range of interests in acoustics and related fields to get together during the year. Each regional chapter provides an ideal chance for new members to learn about the ASA and to gain experience in managing the affairs of the Society. The chapters should encourage not only membership in the Society but attendance at regular meetings as well.

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Technical Specialty Group Report: Computational Acoustics

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At the spring 2018 meeting of the Acoustical Society of America (ASA) in Minneapolis, MN, the Executive Council approved the formation of a new Technical Specialty Group (TSG) in Computational Acoustics (CA). The CA TSG will hold its first official meeting at the fall 2018 meeting in Vancouver, BC, Canada. As indicated in the ASA Rules (Sect. 18), "Technical Specialty Groups are established to organize technical sessions at meetings of the Society in new or evolving acoustical areas not within the scopes of the existing Technical Committees…" Although a TSG functions similarly to a technical committee (TC) in that it organizes sessions, undertakes technical initiatives, and is represented on the ASA Technical Council, it does have significant limitations relative to a full TC, for example, it does not confer awards and medals. After an initial period of three years, the TSG may be renewed, disestablished, or converted into a TC.

Computational acoustics is a well-established yet still rapidly expanding area of acoustics that attracts a broad range of researchers across the spectrum of the current TCs. The effort to establish a CA TSG began in earnest at the Honolulu (HI) meeting in December 2016, when Amanda Hanford (of the Pennsylvania State University Applied Research Laboratory, University Park) and I first circulated a petition and collected signatures. The requisite 50 signatures to form a TSG were eventually gathered before the spring 2018 meeting in Minneapolis. The signatories to the petition listed 10 different TCs as their primary affiliations, with Physical Acoustics, Structural Acoustics and Vibration, and Underwater Acoustics being the most frequently represented.

Along the way to forming the TSG, Amanda and I received encouragement and valuable support from a great many ASA colleagues, whom we will not attempt to list here. Many stimulating discussions helped to sharpen our thoughts about the purpose of a CA TSG and its relationship to the current TC structure. In my view, the primary positive rationale for forming a new CA TSG is well captured by the following quote, which was brought to my attention by Elaine Moran (Director of Operations of the ASA).

> "We also try to seek out groups who are working in some area of acoustics and try to show them the Society will be of value to them. Over the years there has been a lot of discussion about what should be the scope of the Acoustical Society. I choose to take the position that acoustics and the scope of our Society should include whatever those who call themselves acoustical scientists are doing, which they regard as acoustics."
> *Wallace Waterfall (ASA founding member, 1st Secretary, 1st Editor-in-Chief, 8th Treasurer), Address to the Narragansett Regional Chapter, 1966.*

Because the ASA exists primarily to serve the professional interests of its members, if it can better serve a substantial segment of the membership through the

formation of a new TSG, that would be a suitable and sufficient justification for forming one. Given the increasingly computational nature of acoustics and the sciences in general, and the interest of many members of the ASA in recent developments in this area, it is important that the ASA deepen its support for CA-related activities. Particularly worth noting is that many of the signatories of the TSG petition were relatively young ASA members, which likely reflects the emphasis of their research projects and employment interests. Hence the CA TSG should help the ASA to encourage and retain younger members. Many petition signatories also hail from countries other than the United States and Canada; thus the TSG can help expand the reach of the ASA to researchers in other countries.

Although most of the feedback we received regarding formation of a CA TSG was enthusiastic, some thoughtful arguments were also made against the idea. The most frequent was that computational acoustics is a *tool* that should be discussed within the scope of the existing TCs rather than a proper *research topic* warranting focused discussions in a separate, specialized group. In response, it can be argued that computer science is now widely accepted as a legitimate academic discipline, and in many other fields (e.g., fluid dynamics, physics, and biology), computational techniques are regarded as an important and rapidly expanding area of inquiry, distinct enough to be the subject of specialized groups, meetings, and journals. Thus the formation of a CA TSG is really just a recognition of the important status of computation across the spectrum of modern science.

Currently, CA topics are often discussed independently within many of the TCs. This lack of interaction can be a detriment to scientific progress because many computational approaches have multiple applications. One pertinent example is the formulation of computational methods for sound refraction and scattering, such as the parabolic equation, that have applications both underwater (e.g., Jensen et al., 2011) and in the atmosphere (e.g., Salomons, 2012). Another example is finite-element methods for calculating sound fields in interior spaces, which are important in structural, engineering, and architectural acoustics (Thompson, 2006; Marburg and Nolte, 2008; Vorländer, 2013). The CA TSG thus provides a forum for researchers to discuss recent advances in these and other topics crossing the existing TC boundaries.

The technical scope envisioned for the TSG includes the following topics:

- Numerical methods for acoustic wave propagation, scattering, interactions with structures and boundaries, radiation, and other acoustically related phenomena
- Practical utilization of acoustical computations for engineering and noise control, and integration into other simulations
- Optimization, parallelization, and acceleration of computational algorithms
- Validation, benchmarking, and uncertainty analysis in computational models
- Computational learning methods, data analytics, and visualization

The first of these topics includes various numerical methods for solving differential and integral equations, for example, finite-difference methods, boundary-element methods, finite-element methods, parabolic equations, wavenumber integration, and ray tracing. It lies at the heart of how most researchers view computational acoustics. Some particular problems that have received strong interest in recent years include methods to efficiently handle complex boundaries and irregular meshes (e.g., Thompson, 2006; Marburg and Nolte, 2008), time-domain formulations for attenuation and impedance (e.g., Tam and Auriault, 1996), three-dimensional solutions for sound propagating in the ocean and atmosphere (e.g., Castor and Sturm, 2008), wave propagation and scattering in moving and inhomogeneous media (e.g., Ostashev and Wilson, 2016), and nonlinear wave propagation (sonic booms and explosions; e.g., Blanc-Benon et al., 2002).

Regarding the practical utilization of acoustical computations for engineering and noise control, a vital research area is how to efficiently capture complex physical phenomena with limited computational resources. A prime example is modeling noise in complex urban environments where phenomena such as reflections, shadowing by buildings, scattering and absorption from facades and balconies, and distributed sound sources are all important for accurate sound-level prediction. Modeling of such phenomena in large urban spaces through direct numerical methods (e.g., finite differences or boundary elements) can be prohibitive even with supercomputers, although such models can be used to calibrate less intensive empirical and heuristic models.

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The remaining topics in the preceding list might be regarded as activities that support, enhance, or leverage development of conventional numerical methods. The TSG can play a valuable role, for example, by promoting technical exchange between researchers on computational methods and best practices. This may involve holding special sessions and workshops on recent advances in computational techniques, websites to facilitate code exchange and crowd-based development, formulation of benchmarks (Hornikx et al., 2015), and sharing of experiences with new software tools.

The CA TSG looks forward to fostering interactions with other ASA TCs. For example, in the rapidly growing field of machine learning (data-driven) computational methods, there is a natural overlap with the Signal Processing in Acoustics TC. We also look forward to interactions with other societies that have already established groups in computational acoustics, such as the European Acoustics Association. We hope to encourage a stronger ASA presence at international conferences on computational acoustics and perhaps to play a role in organizing future conferences on computational topics.

Whether computational acoustics is a primary interest or a secondary one, please join us for a meeting of the TSG and consider participating in our activities in this exciting and growing area of acoustical research!

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Acoustical Society of America at the 2018 Intel International Science and Engineering Fair

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The Intel International Science and Engineering Fair (ISEF; bit.ly/1m2gRQG) is a program of the Society for Science and the Public (societyforscience.org) and is the world's largest international precollege science competition. The Intel ISEF can be thought of as the "Super Bowl" of science fairs. Each year, approximately 7 million high-school students from around the world compete in local science competitions. This year, of those, only 1,800 finalists from 75 countries advanced to showcase their research at the 2018 Intel ISEF that took place in Pittsburgh, PA. Each affiliated regional science fair from around the world can send a prespecified number of participants to the Intel ISEF each year. The finalists competed for over $4 million in awards and scholarships.

In addition to the Intel ISEF Grand Awards, more than 60 Special Awards Organizations (SAOs; bit.ly/2MKjDxt) also provide awards. SAOs are corporations, government agencies, universities, or nonprofits that sponsor educational scholarships, cash prizes, summer internships, scientific field trips, equipment grants, and more for finalists at the Intel ISEF. The 1,800 finalists were eligible for both Intel ISEF Grand Awards and SAO Awards. The American Society of America (ASA) began participating in the SAO program in 1978 to promote acoustics specifically and to recognize the accomplishments of students, mentors, and schools. The long-time participation of the ASA in the Intel ISEF recognizes the importance of promoting STEM education and generating interest in acoustics among all students. In addition to the $6,300 in cash prize award money, the ASA contributes $2,500 to the Intel ISEF Award Program to support local, national, and international media coverage. Keeta Jones, the ASA Education and Outreach Coordinator, works with the Intel ISEF staff to organize the involvement of the ASA with the fair. She also makes announcements to local ASA members to ask them volunteer their time and expertise as judges. The judges must have a minimum of six years of related professional experience beyond receiving their BA, BS, or masters degrees; have a PhD, MD, or the equivalent; or be a current doctoral student with more than four years of doctoral-level research experience or who is within one year of doctoral dissertation defense.

The fair took place from May 13 to 19, 2018. The ASA judging team was led by Jeffrey Vipperman and included Jeff Babich and Laurie Heller. Mr. Babich is president and principal consultant at Babich Acoustics, Pittsburgh, a company that focuses heavily on architectural acoustics, noise control, and test/measurement. Dr. Heller is a professor at Carnegie Mellon University, Pittsburgh. Her research focuses on hearing, psychoacoustics, environmental sounds, and signal detection. Dr. Vipperman is a professor at the University of Pittsburgh. His research focuses on active/passive noise control, noise classifiers, structural acoustics, acoustic cloaking, and noise-induced hearing loss (NIHL). Dr. Vipperman, as ASA head judge, presented the awards to the three winners at the Special Awards Ceremony on May 17, 2018.

Those eligible for the ASA cash prize awards were chosen by the three ASA judges from the Intel ISEF finalists. They first searched through the 1,800 titles and abstracts for acoustics relevance (defined in the broadest sense). The relevant projects were then reviewed and cut to the 35 top projects that had acoustics as a primary focus of the research. Most, if not all, of the 13 technical areas of ASA were represented.

The ASA judges then met and discussed the 35 projects, and each judge chose to be a primary or secondary judge for various projects. Next, the primary and secondary judges prejudged the projects by reading the student's poster boards (see **Figure 1**), documentation, and lab notebooks in the absence of the student. From there, the judges selected 20 projects for student interviews. A total of five semifinalists were selected and reinterviewed once or twice more by the entire panel of ASA judges.

After much deliberation, first, second, and third place finalists were chosen and received cash prizes for themselves, their schools, and their mentors. The first place winner received $1,500 plus $200 for the school and $500 for the mentor. The second place finalist received $1,000 plus $100 for the school and $250 for the mentor. The third place finalist received $600 plus $150 cash prize for the mentor. An honorable mention certificate was also awarded. Additionally, the student winners have been invited to attend the upcoming ASA annual meeting in Victoria, BC, Canada. For photos of this and past Special Award Ceremonies, see the Intel ISEF ASA Flickr album at bit.ly/2NKiO3O.

The first place ASA winner was Anwesha Mukherjee (Robotics and Intelligent Machines Category). Ms. Mukherjee is a 10th grader at Westview High School, Portland, OR. She was mentored by teacher Debbie Cooper with a project entitled "A Novel Approach to Recognize Emotion from Speech Using Machine Learning Algorithms to Aid Social Interaction of Kids with Autism." The goal of this project was to aid autistic children in their development of empathy and social interactions by using cues from speech to help compensate for difficulties in reading facial expressions. Ms. Mukherjee's contribution over previous work was developing a heuristic weighting of the mel-frequency cepstral coefficients (MFCCs) that improve classifier accuracy by an additional 3-12%. She investigated a number of machine-learning algorithms and found that the multinomial logistic regression provided the most accuracy for her metrics. Her system was trained and tested using a database of expressive speech. See her full project abstract at bit.ly/2oSSj1H. Ms. Mukherjee also received the Intel ISEF Best of Robotics and Intelligence Category Second Place Award.

Figure 1. *Poster session at the Intel International Science and Engineering Fair.*

The second place ASA winner was Sharmi Shah, an 11th grader from Colonia High School, Colonia, NJ, mentored by teacher James Danch (Physics and Astronomy Category). Ms. Shah's project was "Speech Intelligibility Analysis of Sound-Modulated Laser Signal Countermeasures." It is well established that laser light can be used as a spying device by measuring the modulation of the reflected light from a window. Ms. Shah investigated two coatings for their ability to diffuse the laser light and prevent eavesdropping while maintaining optical clarity. She evaluated the two coatings plus a control by using speech-recognition software to analyze speech recorded by the laser. Silica nanoparticle-epoxy residue reduced intelligibility of the chosen speech by 47% while nanoparticle-dimethylsiloxane fully precluded speech intelligibility by the algorithms. Although the coatings appeared transparent, transmission spectra were measured for the control and coated samples and found that the light transmission was very minorly impacted. The project abstract is available at bit.ly/2wN4S2W. Ms. Shah also received the US Air Force First Place Award in Physics and Astronomy, the National Security Agency (NSA) Research Directorate Physical Sciences First Place Award, and the Intel ISEF Best of Physics and Astronomy Category Fourth Place Award.

An 11th grader, Gabrielle Liu (Systems Software Category), from Ravenwood High School, Brentwood, TN, and mentored by teacher Peter Lowen, took third place. Ms. Liu's project, "Preventing Domestic Violence Using Emotion Recognition in Speech" also used machine learning to monitor emotions but as a screen to pick up domestic violence. She envisions incorporating her algorithms in the burgeoning prevalent use of artificial intelligence personal assistants.

They would serve as a "risk barometer" by monitoring the emotional state of a household in real time. As such, domestic violence could be detected in its early phases. Ms. Liu developed a tiered neural network architecture based on nontext prosodic, acoustic, statistical, and spectral features of speech and then evaluated her algorithms using a library of spoken speech with emotion. The project abstract is available at bit.ly/2NQkCsb. Ms Liu also received the Samvid Education Foundation's Geno Third Place Award.

Honorable mention went to Rohan Ahluwalia (Biomedical Engineering Category), also from Westview High School and also mentored by teacher Debbie Cooper. Mr. Ahluwalia's project was "A Fully Functional Closed- Loop System Using Ultrasound Imaging to Automatically Detect Lipohypertrophy in People with Insulin-Dependent Diabetes." Lipohypertrophy is an accumulation of fat under the skin that occurs in diabetics who self-administer multiple daily injections. Early detection is key to preventing a much more painful and unsightly condition. Mr. Ahluwalia developed an edge-detection algorithm that can discern early-stage lipohypertrophy using ultrasound images that cannot be otherwise detected by sight or feel. Using a database of ultrasound images, his algorithm was determined to be correct 85% of the time. Human trials of his device also proved successful. Mr. Ahluwalia abstract is available at bit.ly/2wPbRHF.

The judges look forward to the many great accomplishments and leadership from the many talented youth who participated in the Intel ISEF.

"Listen Up! And Get Involved!"

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The first goal of the Women in Acoustics (WIA) Committee (WIAC) of the Acoustical Society of America (ASA; Isakson, 2010; Ronsse, 2017) starts "to consider the special problems of attracting women to acoustics at all levels of the educational process…." (See womeninacoustics.org for a complete list of the WIA goals.) In 2008, the WIAC began considering possible outreach activities that its members could do at each national ASA meeting to help local girls. Because the daughters of some of the WIAC members participate in local troops, the Girl Scouts of America was selected as the primary recruitment venue for an acoustics outreach session called "Listen Up! And Get Involved!" These events have now been held with Girl Scouts at 10 national meetings and with the Girl Guides at 2 meetings in Canada.

The "Listen Up! And Get Involved!" session was designed to meet the following goals.

1. Give girls the opportunity to interact with female scientists, engineers, and professors. By giving girls the chance of seeing and working with these women role models, the girls may start to think about pursuing such paths for themselves.
2. Expose girls to the wide range of careers that use acoustics. Most girls have a limited perception of available careers and thus have not considered many of the opportunities that are encompassed by the ASA.
3. Allow the girls to feel the excitement of science as they discover and develop an understanding of basic acoustic principles such as the properties of waves, the power of resonance behavior, and how these apply to their everyday experiences.
4. Promote the girls' confidence as they successfully conduct small experiments and explore the equipment used to analyze sound.

Since its inauguration at the Spring 2010 ASA Meeting in Baltimore, MD, 12 "Listen Up! And Get Involved!" sessions have relied heavily on the outstanding support of ASA volunteers. Although the volunteers have been primarily members of WIAC, professional and student ASA members (both men and women) from all technical areas have assisted in making these outreach sessions successful. The volunteers have shared their enthusiasm with the girls as they have engaged in conversations about how everyday experiences with sound are part of the larger world of acoustics and the potential for a career exploring these amazing phenomena.

The volunteers help the girls explore topics, which include some physical principle of sound such as resonance, but most of the stations center on real-world applications such as speech, hearing, musical instruments, and echolocation. A full list of the stations and demonstrations is provided by Vongsawad et al. (2014). To offer all 12 topic areas requires a minimum of 15 volunteers. In addition to the ASA-provided demonstrations, members regularly bring their own materials to share specific aspects of their research and interests. Examples of past member-supplied demonstrations include a ripple tank, a soundscape ecology demonstration, whale songs, acoustic levitation, and active noise control (**Figure 1**). (Con-

Figure 1. Acoustical Society of America (ASA) member Bonnie Schnitta discussing noise cancellation with Girl Scouts at the 173rd meeting in Boston, MA.

tact Keeta Jones if you have a demonstration you would like to bring to one or more of the upcoming meetings.)

The format for these outreach sessions has evolved over the years. For many years, Uwe Hansen, as part of the Education in Acoustics Committee, organized acoustics outreach at the national meetings, in which a middle-school class would attend his hands-on session. The initial Girl Scout outreach sessions (started in 2010) built on Uwe's work. Around 2012, the WIAC joined forces with the Education in Acoustics Committee to revamp both of the outreach sessions (see Bradley, 2015 for more Education in Acoustics outreach efforts). Steps were taken to strengthen the pedagogical approach, update the equipment, and increase the efficiency of both outreach sessions (see Vongsawad, 2014 for details). Since 2013, the two outreach sessions have had the same format, discussed below. Currently, ASA Outreach and Education Coordinator Keeta Jones (2017) works with the WIAC to organize the "Listen Up! And Get Involved!" session, held one evening during the national meeting, and with the Education in Acoustics Committee to organize the daytime session for middle-school students.

As with any outreach event, publicity is required to find attendees. As "Listen Up! and Get Involved!" is primarily marketed toward Girl Scouts between the ages of 12 and 15, the local Girl Scout council is contacted at least 6 months before an upcoming ASA meeting to obtain approval and enlist their aid in spreading the word to local troops. When possible, ASA members who have local Girl Scout connections also help out. At the past two meetings, after the local Girl Scout chapters have registered for the session, additional invitations have been sent to home-school networks, after-school programs, and family event calendars to extend the reach of the event to more families in the area. The goal is to have 50 attendees. Once we have contacted these various groups, ASA member volunteers are recruited to run the demonstrations.

The "Listen Up! And Get Involved!" session involves more than interactive demonstrations. As the guests arrive, we play an entertaining, but homemade, video to get everyone excited and thinking about the sounds in their lives. After the video, our volunteers briefly introduce themselves by saying what area of acoustics they work in before the attendees visit each station in groups of 3 or 4 to spend about 5-7 minutes with each demonstrator. The session closes with an informal Q&A panel about education and careers in acoustics. Participants also have a pizza lunch or dinner for the daytime and evening sessions, respectively. The overall response to the workshops has been very enthusiastic.

Over time, the "Listen Up! And Get Involved!" outreach session will continue to evolve. The Girl Scout local council will still be contacted and Girl Scouts encouraged to come, but the session will no longer be marketed only toward girls. Although it is true that women continue to be underrepresented in the sciences, it doesn't mean that we should deny the opportunity to learn for other underrepresented groups and interested parties. As an example of things that will likely be modified in the spirit of inclusion is the Girl Scout patch (see **Figure 2**), a kind of reward for participating because Girl Scouts tend to collect patches for all their activities. However, the presentation of a "Girl Scout" patch at the end of the outreach session can make other participants feel like they perhaps should not have attended. As such, the WIAC will consider replacing the patch with something more universally collected, such as enamel pins.

Beyond reaching out to other specific underrepresented groups, it is also critical that we continue to maintain the quality of the stations. The stations have remained largely unchanged for seven years. Some equipment has been replaced, but ideas for upgrading the stations should be explored. For example, the voice and hearing stations could include anatomical modals instead of laminated prints of the speech and hearing mechanisms. Another way to improve the educational component of the demonstrations is to add topics and align them to current educational system stan-

Figure 2. *ASA member Sarah Marley holding the "Listen Up! and Get Involved!" patch.* ***Inset****: detail of the patch.*

dards. When we consider how to improve the demonstrations, it is also critical to think about how we can support our volunteers in how to use the demonstrations.

Although many of the session volunteers are veteran demonstrators, just as many have never done science outreach or informal education or have not performed our specific set of demonstrations. As such, it would great to provide mini-informal education-training sessions for volunteers. This way, everyone would have practice and experience with ASA-specific stations and even have a chance to move outside their comfort area by trying new demonstrations. One idea is to offer this training during every meeting between the daytime and evening outreach sessions. This could be open to all meeting attendees, not just our session volunteers. This way, we can send meeting attendees home with the ability to do acoustics outreach at their home institutions.

Ideas for the continued implementation and improvement of the acoustics outreach sessions will be discussed at upcoming WIAC meetings. We hope to continue finding ways to increase the efficacy of these outreach efforts to open the minds of youth to the amazing world of acoustics thanks to the wonderful support of ASA members.

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Obituary | Robert Hickling | *1931-2017*

Robert Hickling, Fellow of the Acoustical Society of America (ASA), passed away peacefully in his home in Huntington Woods, MI, on October 16, 2017, shortly before his 86th birthday. Bob was a retired technical fellow from the General Motors Research Laboratory, Warren, MI; a retired professor from the University of Mississippi, Oxford; and an independent acoustic consultant.

Born on October 28, 1931, in Bologna, Italy, Bob grew up in Scotland and graduated with a MA degree in pure and applied mathematics from the University of St. Andrews, Scotland, UK. He then worked as a scientific officer in the Royal Naval Scientific Service, Teddington, UK, and subsequently transferred to the Underwater Detection Establishment, Portland, Dorset, UK, where he worked on classified sonar targets. Later, he attended the California Institute of Technology, Pasadena, CA, where he graduated with a PhD in engineering science, specializing in computer analysis of problems in underwater acoustic scattering and cavitation bubble collapse and intensity using numerical simulation and experimental methods, resulting in numerous publications.

In 1965, Bob joined the General Motors Research Laboratory, where he initially worked on light scattering by liquid droplets, head injury criteria, and focused pulsed lasers as well as his favorite topics of underwater acoustic scattering and cavitation. In 1971, he formed an acoustic research group that produced many innovative techniques at General Motors such as the first automotive applications of the newly developed Fourier analyzers and experimental modal analysis; the two-microphone acoustic intensity and impedance-tube measurement techniques; air bag design, deployment, and noise; tire noise measurement; and numerical methods for acoustic radiation analysis. In 1981, Bob co-organized a symposium at the General Motors Research Laboratory on "Engine Noise: Excitation, Vibration, and Radiation" that included many well-known and distinguished experts. He also worked on the use of ultrasound in manufacturing, diagnosis, and applications and electromagnetic radiation to stimulate engine combustion that resulted in several US patents.

In late 1988, Bob took early retirement from the General Motors Research Laboratory and joined the National Center for Physical Acoustics (NCPA) at the University of Mississippi, where he became associate director for applied research and research professor of engineering. At the NCPA, Bob worked on underwater acoustics as well as insect acoustics to detect pests in agriculture and acoustic communication by ants, in which he measured ant sounds and developed a near-field theory of communication by ants and possibly by other insects. In 1996, Bob retired from the NCPA and continued to provide consulting on sound intensity measurement through his company, Sonometrics, Inc., including the 2016 publication of the book *Sound-Power Flow: A Practitioner's Handbook for Sound Intensity* that summarizes many of his accomplishments.

Bob was also a Fellow of the American Society of Mechanical Engineers, the Institute of Noise Control Engineering and the Society of Automotive Engineers. He is survived by his three children, Rebecca, Nathan, and David and will be truly missed by his close colleagues from General Motors and the NCPA.

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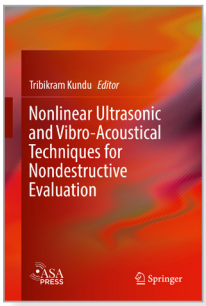


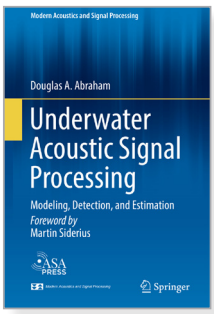
Nonlinear Ultrasonic and Vibro-Acoustical Techniques for Nondestructive Evaluation

Editor: Tribikram Kundu
Copyright: 2019
Publisher: Springer International Publishing
Hardcover: ISBN 978-3-319-94474-6
Edition Number: 1
Number of Pages: XIV, 759
Number of Illustrations and Tables: 138 black and white illustrations, 323 color illustrations
Topics: Engineering Acoustics

- Represents the first book on nonlinear acoustical techniques for NDE applications
- Emphasizes applications of nonlinear acoustical techniques
- Presents the fundamental physics and mathematics behind nonlinear acoustical phenomenon in a simple, easily understood manner
- Covers a variety of popular NDE techniques based on nonlinear acoustics in a single volume

This multi-contributed volume provides a practical, applications-focused introduction to nonlinear acoustical techniques for nondestructive evaluation. Compared to linear techniques, nonlinear acoustical/ultrasonic techniques are much more sensitive to micro-cracks and other types of small distributed damages. Most materials and structures exhibit nonlinear behavior due to the formation of dislocation and micro-cracks from fatigue or other types of repetitive loadings well before detectable macro-cracks are formed. Nondestructive evaluation (NDE) tools that have been developed based on nonlinear acoustical techniques are capable of providing early warnings about the possibility of structural failure before detectable macro-cracks are formed. This book presents the full range of nonlinear acoustical techniques used today for NDE. The expert chapters cover both theoretical and experimental aspects, but always with an eye towards applications. Unlike other titles currently available, which treat nonlinearity as a physics problem and focus on different analytical derivations, the present volume emphasizes NDE applications over detailed analytical derivations. The introductory chapter presents the fundamentals in a manner accessible to anyone with an undergraduate degree in Engineering or Physics and equips the reader with all of the necessary background to understand the remaining chapters. This self-contained volume will be a valuable reference to graduate students through practising researchers in Engineering, Materials Science, and Physics.

About the Editor | Tribikram Kundu is a Professor in the Department of Civil Engineering and Engineering Mechanics at the University of Arizona. Dr. Kundu has made significant and original contributions in both basic and applied research in nondestructive testing (NDT) and structural health monitoring (SHM) by ultrasonic and electromagnetic techniques. His fundamental research interests are monitoring the health of existing and new structures by ultrasonic and other NDT techniques. His research requires knowledge of elastic wave propogation in multi-layered solids, fracture mechanics, computational mechanic, geo- and biomechanics. He has collaborated extensively with international and U.S. scientists. He has spent 28 months in the Department of Biology, J.W. Goethe University, Frankfurt, Germany, first as an Alexander von Humboldt Fellow and then as a Humboldt Research Prize winner. He is a Fellow of the Acoustical Society of America.



Underwater Acoustic Signal Processing

Modeling, Detection, and Estimation

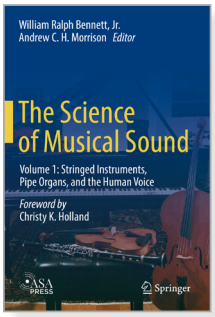
Author: Douglas A. Abraham
Series: Modern Acoustics and Signal Processing
Copyright: 2019
Publisher: Springer International Publishing
Hardcover: ISBN 978-3-319-92981-1
Series ISSN: 2364-4915
Edition Number: 1
Number of Pages: XXXII, 826
Number of Illustrations and Tables: 76 black and white illustrations, 113 color illustrations
Topics: Signal, Image and Speech Processing

- Offers a balanced presentation of the theory and application of signal detection in underwater acoustics
- Addresses a broad audience of practicing sonar engineers, students, and researchers in underwater acoustic signal processing

- Includes relevant background material on underwater acoustics, sonar systems, signal processing, and statistics
- Provides an accessible reference for sonar engineers developing detection algorithms for underwater acoustic sensing systems
- Suitable for use in graduate-level courses in sonar signal processing

This book provides comprehensive coverage of the detection and processing of signals in underwater acoustics. Background material on active and passive sonar systems, underwater acoustics, and statistical signal processing makes the book a self-contained and valuable resource for graduate students, researchers, and active practitioners alike. Signal detection topics span a range of common signal types including signals of known form such as active sonar or communications signals; signals of unknown form, including passive sonar and narrowband signals; and transient signals such as marine mammal vocalizations. This text, along with its companion volume on beamforming, provides a thorough treatment of underwater acoustic signal processing that speaks to its author's broad experience in the field.

About the Author | Douglas A. Abraham received B.S., M.S., and Ph.D. degrees in electrical engineering and an M.S. degree in statistics from the University of Connecticut, Storrs. He has performed basic and applied research in underwater acoustic signal processing at the Naval Undersea Warfare Center (New London, CT), the NATO SACLANT Undersea Research Centre (La Spezia, Italy), and the Applied Research Laboratory at Pennsylvania State University. He presently continues his professional and technical activities as a consultant. Dr. Abraham has also taught at the University of Connecticut as visiting faculty, and managed basic and applied research programs at the Office of Naval Research through an intergovernmental personnel assignment.



The Science of Musical Sound

Volume 1: Stringed Instruments, Pipe Organs,and the Human Voice

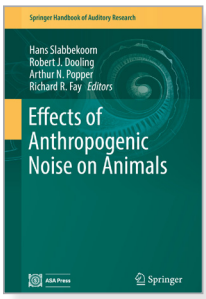
Author: William R. Bennett, Jr.
Editor: Andrew C. H. Morrison
Copyright: 2018
Publisher: Springer International Publishing
Hardcover: ISBN 978-3-319-92794-7
Edition Number: 1
Number of Pages: XXVII, 440
Number of Illustrations and Tables: 185 black and white illustrations, 84 color illustrations
Topics: Acoustics

- Provides unique historical anecdotes relevant to the science
- Includes illustrations and photographs depicting key concepts in novel ways
- Designed for the students who fear math as well as the mathematically inclined
- Written by a world-renowned physicist with a passion for the physics of music
- Covers the basics of oscillations, waves and musical instrument analysis

This textbook is a product of William Bennett's work in developing and teaching a course on the physics of music at Yale University to a diverse audience of musicians and science students in the same class. The book is a culmination of over a decade of teaching the course and weaves together historical descriptions of the physical phenomena with the author's clear interpretations of the most important aspects of the science of music and musical instruments. Many of the historical examples are not found in any other textbook available on the market. As the co-inventor of the Helium-Neon laser, Prof. Bennett's knowledge of physics was world-class. As a professor at one of the most prestigious liberal-arts universities in the world, his appreciation for culture and humanities shines through. The book covers the basics of oscillations, waves and the analysis techniques necessary for understanding how musical instruments work. All types of stringed instruments, pipe organs, and the human voice are covered in this volume. A second volume covers the remaining families of musical instruments as well as selected other topics. Readers without a background in acoustics will enjoy learning the physics of the Science of Musical Sound from a preeminent scientist of the 20th century. Those well versed in acoustics will discover wonderful illustrations and photographs depicting familiar concepts in new and enlightening ways.

About the Author | William R. Bennett, Jr. (1930-2008) was a renowned physicist and professor at Yale University. Prof. Bennett is best known as co-inventor of the Helium-Neon laser. For over ten years, Prof. Bennett taught a widely popular undergraduate course on the physics of music at Yale, upon which this two-volume text is based. Prof. Bennett completed his B.A. at Princeton and received his Ph.D. from Columbia University in 1957. Over the course of his long career, Prof. Bennett was the recipient of numerous awards and honors, and served as master of Yale's Silliman College from 1981-1987.

About the Editor | Andrew Morrison is an Associate Professor at Joliet Junior College in Joliet, IL. He completed his B.S. at the University of Northern Iowa in 2000 and his Ph.D. at Northern Illinois University in 2005. Prof. Morrison is Chair of the Acoustical Society of America Musical Acoustics Technical Committee.



Effects of Anthropogenic Noise on Animals

Editors: Hans Slabbekoorn, Robert Dooling, Arthur N. Popper, and Richard R. Fay

Series: Springer Handbook of Auditory Research

Copyright: 2018

Publisher: Springer-Verlag New York

Hardcover: ISBN 978-1-4939-8572-2

Series ISSN: 0947-2657

Edition Number: 1

Number of Pages: XVIII, 309

Number of Illustrations and Tables: 39 b/w illustrations, 42 color illustrations

Topics: Neurosciences

- Brings together what is known about effects of sound on vertebrates
- Provides a critical introduction to fundamental principles of hearing andacoustics that are needed by all investigators interested in effects of noise on animal
- Chapters focus on taking what is known about basic hearing principles and applying them to the acoustic world of animals

Over the past several years, many investigators interested in the effects of man-made sounds on animals have come to realize that there is much to gain from studying the broader literature on hearing sound and the effects of sound as well as data from the effects on humans. It has also become clear that knowledge of the effects of sound on one group of animals (e.g., birds or frogs) can guide studies on other groups (e.g., marine mammals or fishes) and that a review of all such studies together would be very useful to get a better understanding of the general principles and underlying cochlear and cognitive mechanisms that explain damage, disturbance, and deterrence across taxa. The purpose of this volume, then, is to provide a comprehensive review of the effects of man-made sounds on animals, with the goal of fulfilling two major needs. First, it was thought to be important to bring together data on sound and bioacoustics that have implications across all taxa (including humans) so that such information is generally available to the community of scholars interested in the effects of sound. This is done in Chaps. 2-5. Second, in Chaps. 6-10, the volume brings together what is known about the effects of sound on diverse vertebrate taxa so that investigators with interests in specific groups can learn from the data and experimental approaches from other species. Put another way, having an overview of the similarities and discrepancies among various animal groups and insight into the "how and why" will benefit the overall conceptual understanding, applications in society, and all future research.

About the Editors | Hans Slabbekoorn is an Associate Professor at Leiden University. Robert J. Dooling is a Professor in the Department of Psychology at the University of Maryland. Arthur N. Popper is Professor Emeritus and research Professor in the Department of Biology at the University of Maryland, College Park. Richard R. Fay is Distinguished Research Professor of Psychology at Loyola University Chicago.

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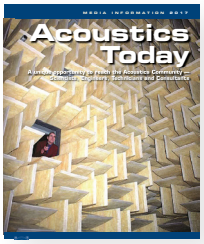

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