5th Mini-Conference of Acoustics (MCA)
6 May 2015

American Center for Physics
College Park, Maryland

Washington DC Chapter of the
Acoustical Society of America

Program

6:00-6:40 Upload Presentations / Ice breaker

6:40-6:45 Welcome and introduction to the Washington DC Chapter of the Acoustical Society of America and the 5th Mini-Conference of Acoustics
Shane Guan (President, Washington DC Chapter of the ASA)

6:45-7:00 Textual and graphical semantic query of sound-speed-profile databases for acoustical training
Daniel T. Redmond, Charles F. Gaumond and Jason E. Summers

7:00-7:15 A parametric ambient-noise model for video-game based education in acoustics*
Valerie Rennoll, Daniel T. Redmond and Jason E. Summers

7:15-7:30 Speaking rate effects on speech perception in adult cochlear implant users*
Brittany N. Jaekel, Rochelle S. Newman and Matthew J. Goupell

7:30-7:45 An improved two-microphone transfer function method for measuring oblique absorption coefficient in the free field using numerical modeling
Hubert Seth Hall, Joseph Vignola, John Judge, Diego Turo and Teresa Ryan

7:45-8:00 Augmenting acoustic phonetics with articulatory features for phone recognition*
Ganesh Sivaraman

8:00-8:15 Listeners' perception of gender after raising vowel formant frequencies in a male speaker*
Betsy Stickels, Sally Gallena and Emily Stickels

8:15-8:30 Time delay spectrometry – A spread spectrum technique for audio, NDT, and medical acoustics
Paul M. Gammell

8:30-8:45 Analysis of signal detection SNR limits in snapshot-deficient scenarios with colored noise*

Jose A. Diaz-Santos and K.E. Wage

8:45-9:00 The broader impact of practicing communication through social media: from Twitter to NSF*

Alexis Blaine Rudd

* Papers qualified for the Best Student Papers Competition

Abstracts

Textual and graphical semantic query of sound-speed-profile databases for acoustical training

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We introduce SSPro, a semantic graphical and textual query engine for sound-speed-profiles (SSPs), and discuss how it supports the automated semantic classification, storage, search, and retrieval of archived and user-defined SSPs. Conventional SSP databases index archived SSPs by location, time of year, and time of day. If SSPs are to be used to simulate an environment with specific physical phenomena—for example, for training operators—this requires expert knowledge of when and where certain SSP attributes occur. In contrast, SSPro uses machine classification to identify semantically meaningful attributes of SSPs, such as “has a surface duct” or “contains a deep sound channel,” and estimate parameters associated with these parameters, such as “layer depth” or “cutoff frequency” using a linearized MacKenzie parameterization of the SSP [K. V. MacKenzie, J. Acoust. Soc. Am., 70, 807–812 (1981)]. The query engine enables users to specify desired attributes of the SSP with natural-language labels and uses these labels to search the ontological database for corresponding SSPs and/or times/locations associated with the desired SSP. The search capability is extended by a graphical user interface that allows users to draw an SSP or manipulate an existing SSP to visually represent the desired search attributes.

A parametric ambient-noise model for video-game based education in acoustics*

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An ambient noise model was developed for inclusion in the educational video game, WaveQuest. Computational noise models such as DANM [Leigh and Eller, APLUW TM 206 (2006)] present a significant computational load. But, as WaveQuest aims to educate users on the fundamental concepts of underwater acoustics, it was important that the game clearly demonstrate the main sources of noise in the sea and their impacts on sonar performance with accurate phenomenological dependence on environmental variables. Due to the variability of noise sources based on time, location, and depth, it is difficult to develop both an accurate and complete parametric model. Here, the ambient noise model was developed based on existing research to include five major sources seismic, shipping, wind, rain, and thermal noises. Each of these noise types was modeled parametrically with respect to frequency and depth. Shipping, rain, and wind noise models were also developed to include dependence on shipping level, wind speed, and rain rate, respectively. To estimate the overall noise level, each of these noise sources were summed incoherently. Overall, the model was found to give reasonable predictions when compared to available data and illustrated with sufficient fidelity those ambient-noise concepts important to WaveQuest.

Speaking rate effects on speech perception in adult cochlear implant users*

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Speaking rate varies across and within talkers, contributing to acoustic variability in the speech signal. For the listener, one effect of speaking rate is changes in perception of phonemes, i.e., the short, meaningful speech sounds that are the building blocks of language. Speaking rate changes can affect the durations of certain phonemes (Crystal & House, 1982), and listeners must "rate normalize" to properly identify words (Miller, 1981). While normal-hearing listeners appear to rate normalize almost automatically (Newman & Sawusch, 2009), it is unknown if people with profound hearing loss, who use auditory prostheses called cochlear implants (CIs), can adjust their perception of phonemes to account for speaking rate. The CI's purpose is to encode speech signal information onto the impaired person's auditory nerve, but because of processing constraints, the signal is quite degraded—thus, rate normalization could be affected. CI users listened to sentences spoken at different rates and identified the final word heard, of which the first phoneme varied. CI users were able to rate normalize, with the same acoustic stimulus being heard as different phonemes depending on the preceding sentence's rate. This indicates speaking rate information can be perceived and applied to degraded input from a CI.

An improved two-microphone transfer function method for measuring oblique absorption coefficient in the free field using numerical modeling

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The use of the two-microphone free field transfer function technique has remained unchanged since its development. In practice, finite sample sizes limit measurement fidelity due to edge effects. The sound field contribution from edge diffraction has generally restricted the technique from use at low frequencies (< 300 Hz) and required test panels greater than several square meters in area. This effort intends to characterize the diffraction contribution to the sound field for relatively small panels. Using numerical modeling, samples of like acoustic properties were excited by a point source at normal incidence to quantify the diffraction term. Following validation via comparison with impedance tube data, the diffraction term was incorporated in an updated derivation of the complex reflection coefficient equation and validated experimentally for a known reference material at both normal and oblique incidence. The goal of this work is to validate measurements of oblique absorption coefficient in an anechoic chamber for panels smaller than one square meter at frequencies down to 100 Hz.

Augmenting acoustic phonetics with articulatory features for phone recognition*

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In articulatory phonetics, a phoneme's identity is specified by its articulator-free (manner) and articulator-bound (place) features. Previous studies have shown that acoustic-phonetic features (APs) can be used to segment speech into broad classes determined by the manner of articulation of speech sounds; compared to MFCCs, however, APs perform poorly in determining place of articulation. This study explores the combination of APs with vocal Tract constriction Variables (TVs) to distinguish phonemes according to their place of articulation for stops, fricatives and nasals. TVs were estimated from acoustics using speech inversion systems trained on the XRMB database with pellet trajectories converted into TVs. TIMIT corpus sentences were first segmented into broad classes using a landmark based broad class segmentation algorithm. Each stop, fricative and nasal speech segment was further classified according to its place of articulation: stops were classified as bilabial (/P/, /B/), alveolar (/T/, /D/) or velar (/K/, /G/); fricatives were classified as labiodental (/F/, /V/), alveolar (/TH/, /DH/, /S/, /Z/), palatal (/SH/, /ZH/) or glottal (/HH/); and nasals were classified as bilabial (/M/), alveolar (/N/) or velar (/NG/). Polynomial kernel support vector machines were trained on APs concatenated with vocal tract constriction features. Results showed that combining acoustic and articulatory features leads to reliable recognition of manner and place of articulation, and improves phone recognition.

Listeners' perception of gender after raising vowel formant frequencies in a male speaker*

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Current clinical practice in voice therapy for male-to-female (MtF) transgender clients focuses on raising modal frequency to a level that can be representative of either male or female, termed "gender neutral". Listening studies have revealed that raising pitch alone is not sufficient to change gender perception. One possible explanation is that the vowel formant frequencies (F1-F3) have remained in a male range. Literature states, MtF transgender clients can raise their vowel formant frequencies by altering tongue position and lip posture but does not specify which vowel formants or at which level is adequate to change gender perception. The purpose of this study was to identify which formants and frequency levels are most influential in listeners' perception of gender. Samples were manipulated in MatLab by isolating each formant using a corresponding band pass filter, modifying the isolated formant, and then combining the modified and unmodified components. Vowel formant frequencies were increased in 20% increments. Samples were randomized into a video format and played for listeners to categorize each sample as male, female, or gender neutral.

Results of this study indicate an increase in listeners' perception of not-male voice when the entire vowel formant envelope is raised maximally.

Time delay spectrometry – A spread spectrum technique for audio, NDT, and medical acoustics

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Time Delay Spectrometry (TDS) is a spread spectrum technique that produces the spectral response of a system for a prescribed range of time delays. A Fourier transform of the resulting spectra provides the full analytic signal representation of the time domain response within the selected time window.

The theoretical basis of this technique is explained along with how the measurement parameters of sweep rate and bandwidth control the resolution and the signal-to-noise (S/N) improvement. A few practical applications in audio engineering, nondestructive testing (NDT), and medical equipment are explained. In one NDT application the S/N improvement enabled a measurement with a time/frequency resolution product smaller than expected from the commonly-quoted expression, enabling assessment of bonds that could not be inspected by other techniques. A few commercial NDT systems are currently available.

A simple TDS system that has been used to calibrate miniature hydrophones used in biomedical ultrasound is explained. This system can be constructed using equipment that is available in most laboratories, the key items being a swept frequency source and an adjustable low frequency filter, along with a few inexpensive commercial components such as frequency mixers..

Analysis of signal detection SNR limits in snapshot-deficient scenarios with colored noise*

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Most source enumeration algorithms assume a white noise background, however in many underwater applications the background noise is colored. Various researchers have investigated the problem of source enumeration when the background noise is colored. The standard approach is to apply a whitening filter before the signal enumeration algorithm. Most of these algorithms perform poorly when the number of noise snapshots used to estimate the whitening filter is small. Nadakuditi and Silverstein [IEEE J. Sel. Topics Signal Process., 2010] developed an algorithm for source enumeration using random matrix theory that provides the fundamental SNR limits for snapshot-deficient scenarios with arbitrary noise. Nadakuditi and Silverstein's analysis focuses on the performance when the number of signal plus noise snapshots varies while the number of snapshots used for the whitening filter stays constant. This talk analyzes the performance of this algorithm when number of snapshots available to estimate the whitening filter varies from N to 10 N. Simulations using two different colored noise models for a large vertical linear array located in the deep ocean will be presented.

The broader impact of practicing communication through social media: from Twitter to NSF*

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Available research funding has decreased in line with the Budget Control Act of 2011, which resulted in funding cuts across both defense and non-defense discretionary programs. These cuts have had a negative impact on many researchers, including those at the Acoustical Society of America. Legislation and appropriations by congress have a direct effect on the funding levels and research priorities of federal agencies. Clear and relatable communication of these impacts is important to both the public and the members of the US Congressional committees, most of whom do not have a background in scientific research. Consideration of the technical and educational background of the audience is vital to clear communication, and agencies such as the National Science Foundation (NSF) are placing increasing emphasis on the broader impacts of scientific proposals and how research will benefit the people of the United States. Social media is an opportunity for scientists to get real-time feedback on science communication and to practice translating scientific jargon for an audience of non-specialists and explaining technical concepts succinctly (often in 140 characters or less).