Session 4aPP: Psychological and Physiological Acoustics
Open Source Audio Processing Challenge Results—Hackathon Challenge Presentations

Odile Clavier, Chair Creare LLC, 16 Great Hollow Rd, Hanover, NH 03755

4aPP1. Speech envelope enhancement to improve cocktail-party listening.

Daniel Cardosi (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, dcardosi@bu.edu), Lucas Baltzell, and Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

A novel speech enhancement scheme is being developed in our lab with the goal of improving speech intelligibility in “cocktail party” listening environments. By manipulating the temporal envelope to increase the salience of acoustic onsets, the algorithm improves access to binaural cues sampled at these onsets. The hope is that this will lead to a more robust spatial perception and improved streaming of competing talkers that could impact a variety of applications including hearing aids and cochlear implants. The algorithm is designed to run in real time, and in order to test its efficacy in real-world environments, it must be implemented on a portable device. For this project, the algorithm is being written in the Teensyduino software and uploaded to a Tympan Rev E DSP unit fitted with a Tympan AIC CODEC daughterboard. External speech is both captured by and played back through a pair of Tympan BTE earpieces, and all processing is performed independently for each ear by the Tympan’s Teensy 4.1 microcontroller. Assuming that the algorithm is successfully implemented, speech signals delivered to the listener will have perceptibly higher crest factors (i.e., higher transient energy) than at input. The portable, bilateral, real-time implementation developed here will allow us to evaluate any associated binaural effects and their consequences in complex real-world listening tasks.

4aPP2. Influence of number of hearing aid compression channels on spatial release from masking.

Marc Brennan (Special Education and Communicative Disord., Univ. of Nebraska-Lincoln, 4075 East Campus Loop South, Lincoln, NE 68583, Marc.Brennan@unl.edu) and Ava Feller (Special Education and Communicative Disord., Univ. of Nebraska-Lincoln, Lincoln, NE)

Listeners with NH take advantage of differences in the spectrum of speech and noise, dips in the noise level, and spatial separation between sound sources to improve their understanding of speech. Listeners with SNHL are less able to take advantage of these differences. Possibly by not considering the influence of access to spectral information under conditions of amplification, hearing-aids only provide marginal improvement in complex listening environments. This experiment examined whether modifying the number of compression channels can improve spatial release from masking (SRM). We obtained sentence recognition for AzBio sentences presented from in front of each participant. Noise that varied temporally and spectrally over time was presented in a collocated condition and in a spatially separated condition. Test conditions included unaided and binaural amplification with an open-source hearing aid. Four and 16 channels of compression were tested. These experimental conditions were based on prior data that documented improved measures of spectral resolution with 16 relative to 4 channels of compression. We hypothesized that SRM would be greater for the condition with 16 channels of compression, relative to the condition with 4 channels of compression. SRM was hypothesized to be poorest for the unaided condition.

4aPP3. Measuring hearing aid compression algorithm preference with the Tympan.

Jennifer Lentz (Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu), Donghyeon Yun (Indiana Univ., Bloomington, IN), and Stuart Smiley (5am Solutions, Bloomington, IN)

In this Tympan challenge, we aim to evaluate listeners' preferences for different Wide Dynamic Range Compression (WDRC) settings in a real-world environment. While WDRC is used to provide comfort for high-level sounds and audibility for low-level sounds, the nonlinear algorithm distorts the stimulus envelope and can alter the input signal-to-noise ratio (SNR). To date, most measurements of speech understanding and preference under different WDRC parameters (e.g., compression speed and compression ratio) have used pre-processed stimuli presented in a sound-attenuating room. In this study, we plan to measure listeners' preference for different algorithm settings by using the Tympan in real-world environments. The Tympan will be programmed based on the listeners' hearing loss, but the user will be able to switch between algorithm settings immediately in their environment using the associated Smartphone app. Using ecological momentary assessment, we will briefly survey the users about their perceptions as they switch between settings. We also will use the Tympan to save acoustic samples and to measure the acoustic properties of the environment and the hearing output, in an effort to determine the drivers of the preference scores [i.e., amount of distortion, degree of audibility, SNR changes (if possible), etc.].

4aPP4. Directionality characteristics of the Tympan open-source hearing aid and earpieces.

Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, alexan14@purdue.edu)

Directional systems in hearing aids are critical for improving speech understanding in the presence of background noise and competing talkers. Tympan is an open-source research platform for speech processing and hearing improvement. The latest Tympan version (Rev E) includes earpieces with a front and a rear microphone, allowing researchers to explore the effects of different directional settings using a realistic hearing aid configuration. The directionality characteristics of the Tympan earpieces were evaluated using a KEMAR in a semi-reverberant sound field. To demonstrate the functionality of Tympan's directionality features, different software settings were manipulated, including the relative delays and gains between the two microphones. Results will be discussed along with considerations for more advanced directional solutions, such as automatic directional switching, adaptive directivity patterns, binaural signal processing, and beamforming.

4aPP5. Immersive multitalker remote microphone system.

Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois at Urbana- Champaign, 456 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801, corey1@illinois.edu) and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL)

Hearing aids and other listening devices perform poorly in noisy, reverberant environments with many overlapping sound sources. Remote microphones, which are worn by a talker and transmit low-noise, low-reverberation speech directly to a hearing device, can improve intelligibility even in adverse environments. However, most commercial remote microphone accessories can only be used with one talker at a time and remove the interaural cues that listeners use to localize sound sources. Using the Tympan open-source hardware platform, we demonstrate a multitalker remote microphone system that preserves interaural cues and room acoustic effects. A pair of wireless microphones transmits sound from two talkers to the Tympan system. A set of real-time adaptive filters running on the embedded processor match the magnitude and phase of the remote microphone signals to the sound received by the earpiece microphones. The earpiece output has the signal-to-noise ratio of the remote microphones but the spatial cues of the on-ear microphones. Because the remote microphones provide low-noise reference signals, the system can track motion in real time and does not need to perform explicit source localization or separation.

4aPP6. A general-purpose pipeline to interface the Tympan hardware with an external computer.

Danielle Benesch (Ctr. for Interdisciplinary Res. in Music Media and Technol., McGill Univ., Danielle Benesch c/o J_er_emie Voix, _Ecole de Technologie Sup_erieure, 1100 Notre-Dame St. W, Montreal, QC H3C 1K3, Canada, dbenesch@critias.ca), Kocherla Nithin Raj (NSERC-EERS Industrial Res. Chair in In-Ear Technologies, _Ecole de Technologie Sup_erieure, Bangalore, Karnataka, India), and Jeremie Voix (NSERC-EERS Industrial Res. Chair in In-Ear Technologies, _Ecole de technologie sup_erieure, Montr_eal, QC, Canada)

Some audio applications require resource-heavy algorithms which cannot be run on real-time digital signal processors such as the Tympan hardware directly due to memory and processing constraints. These algorithms can, however, run on an external computer (PC), and their outcomes can be relayed back to the Tympan where necessary adjustments can be made in the audio processing elements of the underlying hardware. The proposed pipeline includes a 4-channel audio input, the algorithm running on the PC, the Tympan hardware, and a 4-channel output. The Tympan hardware acquires input audio from the Tympan's onboard and/or external microphones with the I2S protocol. The input data are transmitted to the algorithm running on the external PC via USB/Serial communication by exposing the Tympan as a soundcard interface. The output of the PC algorithm is then sent back to the Tympan, through serial communication, so that the underlying audio processing parameters are adjusted. A possible use case is also discussed: a detection algorithm runs on the PC and serial commands are sent back to Tympan to alter the audio outputs, e.g., by playing masking noise. The pipeline is built such that each element can be used independently, abstracting the interconnection of the pipeline elements.

4aPP8. Open-source baby monitor.

Shane W. Lani (Johns Hopkins Univ., 1454 Catherine St., Decatur, GA 30030, shane.w.lani@gmail.com), Jennifer Cooper, Tyler J. Flynn (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD), Jordan Schleif (DoD, Laurel, MD), Adaleena Mookerjee, and O. H. Ott-Pietrak (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD)

New parents often want to know everything that their new little one is up to, even while they are sleeping. The easiest way to accomplish this task is to install an audio baby monitor, comprised of a baby monitor receiver unit near the child's bed and a parental unit which can be carried about the house. These monitors can alert the parents when their bundles of joy are quite upset, but they can also alert the parents with annoying audio from white noise sound machines, lullaby tunes, as well as transient passing motorcycles and landing airplanes. Too much unwanted noise transmitted to the parental unit could lead to the parent turning down their receiver volume and later missing the true alert of their crying little one. The ideal baby audio monitor alerts the parental unit via real time audio only when the baby needs attention, and does not transmit or amplify the received audio for all other noise sources. This presentation outlines the efforts undertaken to develop and test a smart baby monitor using an open-source audio platform and includes performance comparisons of the developed system to commercially available baby monitors.

4aPP9. Open source audio platform ultrasound dosimeter

Jennifer Cooper (Johns Hopkins Appl. Phys. Lab., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723, jennifer.cooper@jhuapl.edu), Tyler J. Flynn, O. H. Ott-Pietrak, Adaleena Mookerjee (Johns Hopkins Appl. Phys. Lab., Laurel, MD), Jordan Schleif (DOD, Laurel, MD), and Shane W. Lani (Johns Hopkins Appl. Phys. Lab., Decatur, GA)

It is well known that hearing loss can occur from exposure to high intensities of sounds >80 dBA; however there is less information about hearing loss from ultrasonic frequencies. There are many commercial and industrial devices that produce ultrasonic acoustic energy ranging from humidifiers, pest and pet repellent devices, and crowd control devices. Even though there are fewer regulations with regard to ultrasound, there have been documented cases of effects including temporary or permanent threshold shifts from exposure to ultrasonic energy. However, there is not a readily available open source solution for quantifying exposure. In this work, a TYMPAN open source hearing aid development platform is used to develop and test a prototype ultrasonic dosimeter. This dosimeter works like a SPL meter, with the inclusion of the ultrasonic 1/3 octave bands, with an audible alarm emitted when the user happens to be in the presence of ultrasonic energy. From the full data recording of a test covering various environments, several key features of the exposure are highlighted. The presentation will cover the development of the algorithms, testing of the device, and post test analysis. Code will be made available via the open source libraries for future users to build upon.

4aPP10. Spatial acoustic processing with a laser distance sensor using a Tympan device.

Kanad Sarkar (Univ. of Illinois at Urbana Champaign, B10 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801-2447, kanads2@illinois.edu), Ryan M. Corey, and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

We will produce a setup for source localization experiments utilizing laser distance sensors with the Tympan device. The Tympan will allow us to perform on-device post processing, improve the portability of our experimental setup, and potentially reduce the electrical noise in our data. We demonstrate the benefits of this device through localizing an acoustic source through lasers placed at different points in space and comparing the Tympan setup to