Beyond Speech and Language: A Soundscape Approach to Aural Habilitation for the Hearing-Impaired Children

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ABSTRACT

Relative to their peers with normal hearing, children with hearing loss are exposed to an environment with less auditory stimulation. Aural habilitation is a specialized early intervention approach to treatment of hearing loss in children to facilitate the development of speech and language ability through provision of appropriate listening devices. The main objective is for the children with hearing loss to develop age-appropriate oral skills and language ability to be able to communicate freely and participate in the hearing world.

Attributed to effective newborn hearing screening programs, the deleterious effects of hearing loss in children may be minimized through early identification and habilitation, making it possible for them to perceive the acoustic environment similar to their hearing peers. To suite the unique characteristics of early-identified children with hearing loss, a soundscape approach to aural habilitation, focusing not only on the perception of speech sounds but also interaction with the acoustic environment, is introduced in this presentation.

BIOGRAPHY

HSIU-WEN CHANG received the first master's degree in Linguistics from National Chung Cheng University, Chiayi, Taiwan, in 1997, the second master's degree in audiology from Washington University in St. Louis, St. Louis, MO, USA, in 2001, and the Ph.D. degree in Biomedical Engineering from National Yang-Ming University, Taipei, Taiwan in 2012. She is a certified audiologist and has extensive clinical experience in adult and pediatric hearing services. She is currently an Assistant Professor with the Department of Audiology and Speech-Language Pathology, Mackay Medical College, New Taipei City, Taiwan. She performs research into many aspects of audiology. Her research interests include the mobile app development for hearing aids, the electrophysiological evaluation of hearing aid effectiveness in infants and young children, and the development of outcome assessment tools in Mandarin Chinese. She also serves as a Board Member of the International Association of Logopedics and Phoniatrics.

Real-time Sound Visualization Tools for Assisting Awareness to Acoustic Events

Prof. Hideki Kawahara
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ABSTRACT

This talk introduces two interactive and real-time tools for understanding speech communication better. They are SparkNG and YANG straight. SparkNG provides a speech production simulator and a real-time vocal tract shape visualizer with introductory tools for spectral analyses. YANG straight provides a source information visualizer and detailed inspector of the signal structure. It also consists of real-time phase information visualizer. They are open-sourced and accessible from our GitHub repositories.

Speech provides rich side information channels which modify/expand its linguistic contents. While the recent resurgence of machine learning technologies makes speech-based communication with smart machines practical and accessible, these rich side information channels are not well exploited. These introduced interactive and real-time tools provide powerful means to understand this side but essential channels. The acquired understanding will enrich smart machines capable of assisting with these channels.

BIOGRAPHY

Hideki Kawahara is a professor emeritus of Wakayama University Japan. He received Bachelor of Eng., Master of Eng. and Ph.D. in Electrical Engineering from Hokkaido University, Sapporo Japan, in 1972, 1974 and 1977 respectively. In 1977, his professional carrier started as a researcher of the NTT Basic Research Laboratories, Tokyo, Japan. In 1992, he joined Human Information Processing Research Laboratories of Advanced Telecommunication Research Institute International (HIP-ATR), Kyoto, Japan as a department head. From 1997 to 2015, he was a professor of Wakayama University and promoted to an emeritus professor in 2015. From 2015 to 2016, he joined Google UK as a visiting research scientist. He is currently a visiting researcher of Riken, Japan. He received the best paper award of 1998-1999 EURASIP on the underlying principle of STRAIGHT. He served as a Distinguished Lecturer of APSIPA in 2015 and 2016. His research interests are signal processing base on auditory representations, speech perception models, and interactive tools for research and education in speech science.

Loudness Evaluation for Older Adults and Some Application Examples

Prof. Kenji Kurakata
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ABSTRACT

The frequency-weighting "A" is widely used for noise evaluation. It has been demonstrated that A-weighted sound pressure levels show a good correspondence with perceived loudness of various sounds. However, the good correspondence is confined to cases in which young people with normal hearing judge the loudness because the A-weighting curve was determined based on their hearing characteristics. Therefore, it is questionable whether the A-weighted sound pressure level can be a good measure of loudness that older adults perceive.

In my talk, I will introduce a new method of noise evaluation that was developed on the basis of hearing characteristics of older adults and show that their perceived loudness can be estimated better than by using the conventional A-weighting. Some application examples will be presented as a demonstration.

BIOGRAPHY

Kenji Kurakata received his Ph.D. degree from Graduate School of Human Sciences, Osaka University in 1994. He is currently a Professor, Faculty of Human Sciences in Waseda University. He received Prize for distinguished achievements or books in noise control (2016), INCE/J in 2016 and Sato Prize, The Acoustical Society of Japan in 2007. His research interest include psychology of hearing, human interfaces , psychology of music and Rehabilitation science/Welfare engineering.

Perception of Sign Sounds by Younger Hearing Impaired People

Prof. Keiichi Yasu
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ABSTRACT

Several previous research has reported that it is difficult to aware of the auditory signals of daily life if there is hearing impairment. In this study, we investigated what kind of daily auditory signals are difficult to listen to young deaf and hard of hearing people. The participants were fifteen university students between the ages of 19 and 22. The auditory background varied (hearing aids, cochlear implants, bilateral hearing loss). The materials were clean sounds without noise and recorded sounds with noise (recorded in Taiwan and Japan). The materials were presented from the loudspeakers in the classroom. The loudness of sounds was around 75 [dBA] at 1 [m] for the closest person. The participants were asked to write the name of the sound. The degrees of familiarity, confidence, and awareness were also evaluated in the mean opinion score (MOS). As a result, the sound with high familiarity had a higher accuracy rate for naming.

BIOGRAPHY

Keiichi Yasu, Ph.D., is a research associate at the Department of Industrial Information, Faculty of Industrial Technology, Tsukuba University of Technology, Japan. He is interested in neuroscience and behavioral aspect of stuttering, hearing impairment and speech perception.

Evaluation of Environmental Sounds for Elderly People Using Hearing Impairment Simulation

Prof. Toshie Matsui
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ABSTRACT

Environmental sounds often transmit danger around us. It is difficult to evaluate the danger transmitted by sounds for elderly people with age-related hearing loss (ARHL) due to the deterioration of higher-order functions. Using Hearing Impairment Simulation (Hisim), we evaluated the danger of passing car sounds and warning sounds for elderly people with ARHL. The Hisim allows normal hearings to experience the hearing of ARHL. The danger assessment of the passing car sound revealed that the danger of traffic sounds was underestimated in ARHL simulation condition. It was not possible to simulate the hearing loss sufficiently with just the same average sound pressure as in the hearing loss. The pairwise comparison of the speed of passing car sound suggested that ARHL may have caused some changes in speed perception. Another environmental sound, warning sounds are often created by sweeping sine waves. Stimuli for our experiments were created by changing F0, the number of harmonics, and the silent duration between sweeps. As a result of the evaluation, it was found that the danger is more easily transmitted to ARHL listeners by the sweep sound starting at lower F0. The condition of this effective warning sound was different from the sweep sound which was evaluated to be "effective for transmitting the danger" in the experiment for the normal hearing listeners. These experimental results suggest that we need to recapture the transmission of information from environmental sounds in our aging society.

BIOGRAPHY

Toshie Matsui is an Associate Professor at Toyohashi University of Technology in Japan, where she has joined in 2017. She received her B. Mus., M. Mus., and Ph.D. in Musicology from Kyoto City University of Arts in 2000, 2004, and 2010, respectively. Her research interest is auditory peripheral modeling. She is now engaged in a couple of research projects on hearing loss simulation. Her goal is to link researches on auditory psychophysics and more complicated sound recognition, such as speech and music. Since she was major in piano performance until her Master, she has been also interested in the professional performers' perception of music.

Environmental Sounds in Taiwan and Japan

Prof. Hiroko Terasawa
M.S. Haruki Onaka
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ABSTRACT

Both Japan and Taiwan are entering the phase of super-aged and aged societies. We anticipate the increase number of age-related hearing impaired population in both countries. Along with the increase of aged population, we will have much greater number of age-related hearing loss (ARHL) patients. Hearing impairment will be no longer a rare disability, but rather a common social issue. We are aiming at using simulated hearing loss technology to understand the soundscape that elderly people experience. Such understanding will lead to the sound design that is friendly for ARHL patients. We conducted field recording of everyday soundsboth in Japan and Taiwan, and we are organizing more than 200 sound clips from public spaces. We will present the distributions of basic acoustics parameters, such as loudness and spectral centroid, of sounds from Taiwan and Japan, and discuss the future directions of experimental approach.

BIOGRAPHY

Hiroko Terasawa is an assistant professor from University of Tsukuba. She received B.E. and M.E. in Electronic Engineering from University of Electro-Communications in Tokyo, Japan, and M.A. and Ph.D. in Music from Stanford University, USA. Her research interest includes timbre perception and music recognition by challenged listeners. She is the recipient of Super Creator Award from IPA Mitoh project, and the Best Speaker Award from Japan-America Frontiers of Engineering Symposium. She is from Tokorozawa, Saitama, which is famous for producing Sayama green tea.

Haruki Onaka is a Master's program student from University of Tsukuba. His Bachelor's and Master's projects focus on the perception of environmental sounds by elderly people. He is from Kakegawa, Shizuoka, which is famous for producing Shizuoka green tea.

A New Mandarin Deceptive Dialogue Database: Data Acquisition through an Interactive Game

Prof. Yi-Wen, Liu
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National Tsing Hua University, Hsinchu, Taiwan

ABSTRACT

Being able to distinguish the differences between deceptive and truthful statements in a dialogue is an important skill in daily life. Extensive studies on the acoustic features of deceptive English speech have been reported, but such research in Mandarin is relatively scarce. We constructed a Mandarin deception database of daily dialogues from native speakers in Taiwan. College students were recruited to participate in a game in which they were encouraged to lie and convince their opponents of experiences that they did not have. After data collection, acoustic-prosodic features were extracted. The statistics of these features were calculated so that the differences between truthful and deceptive sentences, both as they were intended and perceived, can be compared. Results indicate that differ- ent people tend to use different acoustic features when telling a lie; the participants could be put into 10 categories in a dendrogram, with an exception of 31 people from whom no acoustic indicators for deception were found. Without considering interpersonal differences, our best classifier reached an F1 score of 53.37% in distinguishing deceptive and truthful segmentation units. We hope to present this new database as a corpus for future studies on deception in Mandarin conversations.

BIOGRAPHY

Dr. Liu, Yi-Wen received his B.S. degree in EE from National Taiwan University in 1996, and M.S. and Ph.D. degrees in EE from Stanford University in 2000 and 2006, respectively. He completed four years of post-doctoral training (2006-2010) at the Center of Hearing Research in Boys Town National Research Hospital, Omaha, USA. He joined the Department of Electrical Engineering at National Tsing Hua University in 2010 and established the Acoustics and Hearing Group which focuses on (1) Computer modeling of cochlear mechanics and auditory neurophysiology, (2) digital processing of audio signals. He is presently a member of the Association for Research in Otolaryngology (ARO), the Acoustical Society of America (ASA), and the IEEE Signal Processing Society. Being an enthusiastic lecturer, he has been invited numerous times to give talks on hearing science and audio/speech technologies both domestically and internationally. Dr. Liu was a recipient of NTHU Teaching Award in 2014 and 2017.

Vocal Fold Vibration with Aging and Disordered Voices

Prof. Ken-Ichi Sakakibara
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ABSTRACT

The vocal fold vibration is one of the most important factors to regulate the voice quality. The vocal fold vibration is complex system in the strict sense of the mathematical viewpoints and small differences in physical properties of the vocal folds and laryngeal adjustment result in very different vibratory patterns and, consequently, r different voice quality. In this talk, age and sexual differences of the vocal fold vibratory patterns are analyzed using high-speed digital imaging, and as well as the vocal fold vibratory patterns of several types of pathological voices.

BIOGRAPHY

Ken-Ichi Sakakibara studied mathematics (algebraic geometry) in Kyoto University, and was engaged in computer music research in NTT Communication Science Labs. His current research interests are in vocology, singing voice, speech science, and snow science (especially avalanche risk management).

Automatic Detection and Classification of Voice Disorders Using Deep Neural Networks

Dr. Chi-Te Wang
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Far Eastern Memorial Hospital, New Taipei, Taiwan

ABSTRACT

From a health science perspective, a pathological status of the human voice can substantially reduce quality of life and occupational performance, which results in considerable costs for both the patient and society. Current practice standards recommend laryngeal endoscopy for the accurate diagnoses of voice disorders, which requires well-trained specialists and expensive equipment. In places without sufficient medical resources, and for patients without adequate insurance coverage, correct diagnosis and subsequent treatment may be delayed.

With the progress of computation capability during the past decades, several studies had applied machine learning algorithms to detect vocal pathologies from recorded voice samples. Starting from 2015, we founded a research group collaborating with experts from **National Yang-Ming University , Academia Sinica, Yuan Ze University, and National Taiwan University**. We applied the latest deep neural network (DNN) to differentiate between normal and abnormal voice samples based on Mel frequency cepstral coefficients. The accuracy reached 90.52% to 94.26% on database from Far Eastern Memorial Hospital (FEMH). When applied to another voice disorder database from Massachusetts Eye and Ear Infirmary for validation, DNN achieved the highest accuracy (99.32%) than the other classification algorithms.

Although detecting voice disorders had reached a great success, further classification of voice disorders into several distinct disease categories had rarely been attempted before. Using the comprehensive voice disorder database from FEMH, we first proposed that demographic and symptomatic features can be applied to computerized classification. The accuracy ranged from 74% to 83% among the tested machine learning algorithms.

Based on the above experience, we combined acoustic signals and medical records and proposed a multimodal approach to classify pathological voice samples. By transforming acoustic signals into static supervectors or two stages of DNN, the study results showed a highest accuracy of 87.26% for classification between three common categories of vocal diseases, i.e. glottic neoplasm, phonotraumatic lesions, and vocal paralysis.

On behave of the great research partners, it is my honor to present our research works in this unique conference between Taiwan and Japan. We look forward to receiving comments and suggestions from the experts, as well as opportunities for cooperation in the future.

BIOGRAPHY

Dr. Chi-Te Wang received his MD degree from the National Taiwan University, Taipei, Taiwan, in 2003. From 2003 to 2008, he was the resident and chief resident doctor at the Department of Otolaryngology, Head and Neck Surgery at National Taiwan University Hospital. He joined Far Eastern Memorial Hospital as an attending physician since 2008. He visited Mount Sinai Hospital in 2009 and started his career as a laryngologist since he went back to Taiwan. He received the PhD degree from the Institute of Epidemiology and Preventive Medicine at National Taiwan University in 2014. During his professional carrier, he visited Mayo Clinic (Arizona, 2012), Isshiki voice center (Kyoto, 2015), UC Davis voice and swallow center (Sacramento, 2018), and UCSF voice and swallow center (San Francisco, 2018) for continual exposure on the expertise practice of phonosurgery.

He is currently an Assistant Professor in the School of Medicine at National Taiwan University and University of Taipei, lecturing on voice and swallow disorders. He is also a member of councils on the Taiwan Otolaryngological Society and Taiwan Voice Society. He has a wide clinical and academic interest on different fields, including in-office laryngeal procedures, automatic detection and classification of voice disorders, real time monitoring of phonation, and telepractice of voice therapy. Collaborating with experts from artificial intelligence and signal production, Dr. Wang co-hosted the “Voice Disorder Detection Competition” during the conference of 2018 IEEE International Conference on Big Data, receiving more than 100 groups of participants from 27 countries around the world. The research group had also won the grand prize of Far Eastern Spirit Contest in 2018.

The Deep Learning-based Noise Reduction Technology for Cochlear Implant Recipients

Prof. Ying-Hui Lai
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National Yang-Ming University, Taipei, Taiwan

ABSTRACT

The cochlear implant (CI) is one of the popular assistive listening devices for deaf hearing loss individuals. The CI is surgically implanted electronic devices that provide a sense of sound in patients with profound to severe hearing loss. The considerable progress of CI technologies in the past three decades has enabled many CI users to enjoy a high level of speech understanding in quiet. For most CI users, however, understanding speech in noisy environments remains a challenge. In this talk, I will present the proposed approach, called NC+DDAE, to improve the performance of speech intelligibility in noisy listening conditions for CI recipients. In a series of extensive experiments, we conduct qualitative and quantitative analyses of the NC module and the overall NC+DDAE approach.

Moreover, we evaluate the speech recognition performance of the NC+DDAE NR and traditional single-microphone NR approaches for Mandarin-speaking CI recipients under different noise conditions. The experimental results of the objective evaluation and listening test indicate that, under challenging listening conditions, the proposed NC+DDAE NR approach yields higher intelligibility scores than the two compared classical NR techniques, under both matched and mismatched training conditions. More specifically, when compared to the two well-known conventional NR techniques under challenging listening condition, the proposed NC+DDAE NR approach has superior noise suppression capabilities, and gives less distortion for the essential speech envelope information, thus improving speech recognition more effectively for Mandarin CI recipients. The results of this study suggest that the proposed deep-learning based NR approach can potentially be integrated into existing CI signal processors to overcome the degradation of speech perception caused by noise.

BIOGRAPHY

Dr. Ying-Hui Lai received the Ph.D. degree in the Department of Biomedical Engineering from National Yang-Ming University in 2013. From January 2010 to June 2012, Dr. Lai was as a research and development (R&D) engineer in Aescu technology, Taipei, Taiwan, where he engaged in research and product development in hearing aids. From October 2013 to September 2016, Dr.

Lai was a postdoctoral fellow of the Research Center for Information Technology Innovation at Academia Sinica. Following, Dr. Lai was in the Department of Electronic Engineering, Yuan Ze University as an assistant professor for one year. In August 2017, he joined the faculty of the Department of Biomedical Engineering, National Yang-Ming University, as an assistant professor. His research interests include hearing and speech sciences, assistive devices for hearing and communication, acoustic event detection, biomedical signal processing, and artificial intelligence.

Hearing Impairment Simulator: Its Background and Applications

Prof. Toshio Irino
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Wakayama University, Wakayama, Japan

ABSTRACT

In this talk, I introduce our hearing impairment (HI) simulator and its background.

People with HI often have difficulty understanding speech in multi-speaker or noisy environments (Dubno et al., 1984; Humes and Roberts, 1990; Prosser et al., 1991). With HI listeners, however, it is difficult to specify which stage of auditory processing is responsible for the deficit as they might have multiple processing problems.

The HI simulator was developed to study the problems with normal hearing listeners.

The HI simulator is based on the dynamic, compressive gammachirp (dcGC) filterbank which is based on psychoacoustic measurement of auditory filter and compression using a notched noise (Irino et al., 2013). A user interface was added to the simulator and the ratio of an individual's compressive loss to their linear loss can now be varied for a given audiogram. A new algorithm was introduced to improve processing speed to simulate HI sounds. The HI simulator has been used to measure speech intelligibility and the effect of a loss of compression on syllable recognition and to provide environments for behavioral experiments and lecture courses for speech-language therapists.

BIOGRAPHY

Toshio IRINO has been a professor of Wakayama University since August 2002. He received his B.Eng., M.Eng., and Ph.D. degrees in electrical and electronic engineering from Tokyo Institute of Technology, 1982, 1984, and 1987, respectively. From 1987 to 1997, he was a research scientist (a senior research scientist from 1991) at NTT Basic Research Laboratories. From 1993 to 1994, he was a visiting researcher at Medical Research Council - Applied Psychology Unit (MRC-APU, currently CBU) in Cambridge. From 1997 to 2000, he was a senior researcher in ATR Human Information Processing Research Laboratories (ATR HIP). From 2000 to 2002, he was a senior research scientist in the NTT Communication Science Laboratories. He was also a visiting professor at The Institute of Statistical Mathematics from 2005 to 2008. The focus of the current study is a computational theory of the auditory system.

Prof. Irino is a fellow of the Acoustical Society of America (ASA), a senior member of the Institute of Electrical and Electronics Engineering (IEEE), a member of the Association for Research in Otolaryngology (ARO), International Speech Communication Association (ISCA), the Acoustical Society of Japan (ASJ), and the Institute of Electronics, Information and Communication Engineers (IEICE).

Hearing Scale Test and Ear Scale App for Hearing Screening of School-age Children

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National Yang-Ming University, Taipei, Taiwan

ABSTRACT

The Hearing Scale Test (HST) employs ten stratified hearing scales from S1 to S10 for children, (S1 to S15 for elderly) each hearing scale contains four test tones with adjacent scales differ from each other by 5 dB, ranging from 1dB (S1) to 71 dB (S15). The starting hearing scale of the HST is S5 (1000, 2000, and 4000 Hz at 20dB, and 500 Hz at 25 dB). The HST test reports: scales S1–S5 indicate normal hearing; scales S6 and S7 indicate possible hearing impairment, and scales S8–S15 indicate confirmed hearing impairment. The HST audiometer is designed for early detection of unidentified hearing impairment children, which can replace the routine hearing screening currently and tuning fork test.

Pure tone audiometry is the gold standard to evaluate the hearing level. However, it has to be performed in a sound-proof room by hearing professionals. Therefore, it is not easy accessible and makes daily hearing monitoring difficult for the patients. To solve this main problem, our research team has developed a mobile phone-based application “Ear Scale App”, which contains four test tones with adjacent hearing scales differ from each other by 5 dB, ranging from 1dBHL (S1) to 96 dBHL (S20)

We aim to establish miniature listening cloud monitoring mode and build Ear Scale App for smart phones and tablets.2. Verify the hearing regarding the value of the Ear Scale App in patients with sudden deafness, and hearing regarding the value of average test results with pure tone hearing threshold value differences, to become a self that occupy the home inspection, tracking patients with sudden deafness hearing, can assist to rehabilitation of the application of tools. In this study, We try to develop the IOS application- Ear Scale App and combined with cloud monitoring and hearing detection to verify patients with sudden deafness in the TVGH. Ear Scale App introduction in youtube videos: https://youtu.be/5x6a_KmRjMc.

BIOGRAPHY

Dr. Liao, Wen-Huei (廖文輝) is an Assistant Professor at the National Yang-Ming University, Taiwan. He researches in the field of human sensation “Hearing”, about how to recover and maintain hearing level. Specifically, his research focuses upon sudden sensorineural hearing loss treatment and tinnitus relief. He is also a clinically certified Otolaryngologist and a staff physician at the department of otolaryngology Taipei Veterans General Hospital, Taiwan. He had proposed a method for hearing screening “Hearing Scale Test “by using a quantitative scale, and to test it in school-aged children. He developed the a mobile phone-based application “Ear Scale App”, which contains four test tones with adjacent hearing scales differ from each other by 5 dB, ranging from 1dBHL (S1) to 96 dBHL (S20), making it possible for these patients to monitor and report their hearing level. The Ear Scale App can detect latent hearing loss and have the ability to distinguish the level of hearing. It has been tested and verified in elementary school students. He also develops the IOS application- Ear Scale App and combined with cloud monitoring and hearing detection to verify patients with sudden deafness in the TVGH.