Session 4pPPc

Psychological and Physiological Acoustics: General Topics in Psychological and Physiological Acoustics VI (Poster Session)

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All posters will be on display from 3:40 p.m. to 5:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 3:40 p.m. to 4:30 p.m. and contributors of even-numbered papers will be at their posters from 4:30 p.m. to 5:20 p.m.

Contributed Papers

4pPPc1. A computational model of binaural speech intelligibility level difference. Kalle J. Palomäki (Adaptive Informatics Research Centre, Helsinki University of Technology, P.O. Box 5400, 02015 Espoo, Finland, kalle.palomaki@tkk.fi), Guy J. Brown (University of Sheffield, Dept. of Computer Science, Regent Court, 211 Portobello Street, S1 4DP Sheffield, UK, g.brown@dcs.shef.ac.uk)

This study addresses two questions relating to the binaural intelligibility level difference (BILD). First, we ask whether the BILD is underlain by an equalization-cancellation (EC) mechanism, in which a disparity between the interaural time difference of the target and masker is exploited within each frequency channel, rather than across channels. Second, we consider the effects of three sources of internal noise on the EC mechanism: jitter in neural delays, noise in the equalization process and nonlinearities in the auditory pathway. These issues are investigated using a computational model consisting of peripheral auditory model, binaural processor, auditory scene processor and automatic speech recognition system. The binaural model is based on EC processing, with performance limited by internal noise. The auditory scene processor groups speech harmonics by common F0 and identifies 'glimpses' in which the signal-to-noise ratio is favorable for speech. The performance of human listeners and the computational model are compared on the same speech intelligibility test (Edmonds & Culling, 2005, JASA 117 (5), 3069-3078). The BILD of human listeners can be replicated by adjusting parameters that determine the internal noise in the EC model; however, the speech reception threshold of the model is lower than that of human listeners.

4pPPc2. Bernoulli coding on the auditory nerve and its implications for central auditory processing. Robert A. Houde (Center for Communications Research, 125 Tech Park Drive, Rochester, NY 14623, USA, rahoude@gmail.com), James M. Hillenbrand (Western Michigan University, Dept of Speech Path & Aud., 1903 W. Michigan Ave., Kalamazoo, MI 49008, USA, james.hillenbrand@wmich.edu)

The auditory periphery is well represented as a bank of band pass filter/inner hair cell (IHC) channels, with each IHC providing half wave rectification, amplitude compression, and conversion to firing probability on the auditory nerve (AN) fibers innervating that IHC. Frequency resolution varies dramatically with sound intensity, ranging from sharp tuning near threshold to very broad at high intensities. Cochlear filtering provides a satisfactory representation of broadband characteristics such as timbre but not the fine frequency resolution required for perceptual frequency discrimination. High resolution frequency analysis must, therefore, be provided by post-AN processes. We present a model of AN coding in which fine frequency analysis is carried out at central auditory stages. By this model the stochastic process on each AN fiber resulting from the IHC's firing probability is modeled by a Bernoulli process. As a result, the IHC output signal is transferred to the cochlear nucleus (CN) without further filtering, where it can be recovered by a simple summation over those AN fibers from the region of that IHC. We present a neurally plausible process for narrow band analysis at the CN using the regular pulses of chopper cells.

4pPPc3. The first effect of pitch shift as a function of component spacing. Adam Mielczarek (Acoustics Division, Wroclaw University of Technology, Wybrzeze Wyspianskiego 27A, 50-370 Wroclaw, Poland, adam.mielczarek@pwr.wroc.pl)

The paper presents the results of an experiment regarding the influence of component spacing on the first effect of pitch shift. During the adjustment procedure, the listeners matched the pitch of the three-component complex to the same sensation produced by pure tone. The stimuli were composed of the 3rd, 4th and 5th harmonics of 100, 200 or 400 Hz shifted up in the frequency domain by 30 Hz. The level of each component was 50 dB SPL. The subject was presented with a 5-s sample of the test complex, and after a 500 ms break he had to define the pitch of the three-component complex using the matching tone. The results of the experiment suggest that the pitch shift phenomenon is based on the relative frequency rather than on the absolute frequency or the dominant component number.

4pPPc4. Neuronal representation of pitch ambiguity. Mark Sayles (Centre for the Neural Basis of Hearing, The Physiological Laboratory, Downing Street, CB2 3EG Cambridge, UK, ms417@cam.ac.uk), Ian M. Winter (Centre for the Neural Basis of Hearing, The Physiological Laboratory, Downing Street, CB2 3EG Cambridge, UK, imw1001@cam.ac.uk)

Iterated rippled noise (IRN) is produced by delaying a broadband noise by time d, multiplying by gain g, adding the delayed noise to the original, and repeating this process for n iterations. When $g=+1$ IRN has a well-defined pitch at $1/d$ Hz. If g=-1 the pitch can be ambiguous. A gain of -1 is equivalent to applying a frequency-independent phase shift ϕ of π rads to the delayed noise ($g=+1 \equiv \phi=0$). We recorded spike-trains from single units in the ventral cochlear nucleus in response to IRN with varying ϕ. Units with high best frequencies represented waveform envelope modula-

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tion (independent of ϕ), however, units in the phase-locking range of best frequencies represented stimulus fine structure (which varies with ϕ). Fine structure responders show a gradual transition from a well-defined peak in the interspike interval distribution at d when ϕ=0 to two equal-amplitude peaks flanking d when $\phi=\pi$, and a gradual shift back to a well-defined peak at d as ϕ approaches 2π. Within the dominance region for pitch interspike interval distributions account for psychophysical pitch matches of 1.07/d and 0.94/d Hz for $\phi=\pi/2$ and $3/2\pi$ respectively, as well as the ambiguous pitches of 0.88/d, 1.14/d, and 1/2d Hz heard when $\phi=\pi$ rads.

4pPPc5. The effect of regional dialect on the psychometric reliability and validity of two sets of Mandarin speech audiometry materials. Shawn L. Nissen (Brigham Young University, 138 TLRB, 1190 North 900 East, Provo, UT 84602, USA, shawn_nissen@byu.edu), Richard W. Harris (Brigham Young University, 138 TLRB, 1190 North 900 East, Provo, UT 84602, USA, richard_harris@byu.edu), Jamie Garlick (Brigham Young University, 138 TLRB, 1190 North 900 East, Provo, UT 84602, USA, jamie.garlick@gmail.com), Nathan Richardson (Brigham Young University, 138 TLRB, 1190 North 900 East, Provo, UT 84602, USA, nathan000000@yahoo.com)

Previous research has shown conflicting evidence on the effect of testing an individual's hearing acuity with speech perception materials created in a mutually intelligible, yet non-regional dialect. Thus, the aim of this study is to examine the validity and reliability of using previously developed psychometrically equivalent speech audiometry materials in Mainland Mandarin and Taiwan Mandarin to evaluate the speech perception abilities (word recognition and speech reception threshold) of regional and non-regional listeners of the presented dialects. In addition, this study will investigate whether a native speaker of one Mandarin dialect is able to accurately administer and score the results from listeners of a different regional dialect. Some aspects of the listeners' performance on materials from a non-regional Mandarin dialect were found to be significantly different statistically. However, it is unclear if such differences are large enough to make a difference in the clinical testing of speech perception. In terms of scoring accuracy, a high percentage of agreement was found between the two interpreters from different dialectal backgrounds. [Work supported by research funding from Brigham Young University School of Education]

4pPPc6. Estimating the effective frequency of cochlear implant electrodes using contralateral residual acoustic hearing. Tim Green (UCL, Wolfson House, 4, Stephenson Way, NW1 2HE London, UK, tim.green@ucl.ac.uk), Andrew Faulkner (UCL, Wolfson House, 4, Stephenson Way, NW1 2HE London, UK, andyf@phon.ucl.ac.uk), Stuart Rosen (UCL, Wolfson House, 4, Stephenson Way, NW1 2HE London, UK, stuart@phon.ucl.ac.uk)

For some cochlear implant (CI) users a contralateral hearing aid provides significantly improved speech perception. Important factors in the bimodal transmission of speech spectral information are likely to include the extent to which the frequency selectivity of residual hearing allows additional place-coded channels, and mismatches between frequency-to-place maps across modalities. When acoustic place coding extends above around 500 Hz an overlap of frequency coverage between acoustic and electric hearing may result in interaural conflicts. However, addressing this issue requires accurate knowledge of CI frequency-to-place maps. Effective characteristic frequencies of CI electrodes have previously been estimated using comparisons of the pitch produced by electrical stimulation with that produced by contralateral acoustic sinusoids In the present work, the acoustic stimuli used for pitch comparisons are either sinusoids or 1/3 octave bands of noise. The latter minimize temporal pitch cues and may reduce differences in perceived quality between electrical and acoustical auditory sensations. Electrical stimuli are high-rate (900 pps or greater) single-electrode pulse trains. Comparisons are performed at different levels spaced over the dynamic range and both paired-comparison and adjustment tasks are used. Results will be discussed in relation to speech processing approaches for optimally combining an implant and contralateral hearing aid.

4pPPc7. Performance on auditory temporal-processing tasks for speech and non-speech stimuli by young and elderly listeners. Diane Kewley-Port (Indiana University, Speech and Hearing Sciences, 200 S. Jordan, Bloomington, IN 47405, USA, kewley@indiana.edu), Larry Humes (Indiana University, Speech and Hearing Sciences, 200 S. Jordan, Bloomington, IN 47405, USA, humes@indiana.edu), Daniel Fogerty (Indiana University, Speech and Hearing Sciences, 200 S. Jordan, Bloomington, IN 47405, USA, dfogerty@indiana.edu), Dana Kinney (Indiana University, Speech and Hearing Sciences, 200 S. Jordan, Bloomington, IN 47405, USA, danakin@indiana.edu)

Results from three auditory tasks are presented from a larger series of temporal-processing tasks completed in three sensory modalities by young and older adults. The first task measured temporal gap detection in noise bands. The second and third tasks used digitally processed vowels in four words (pit, pet, pot, put) as the stimuli. The second task required listeners to identify the order of either two- or four-vowel sequences presented monaurally or dichotically. The third task measured the identification of these four vowels when presented either before or after a noise or vowel-like masker (forward- or backward-masking tasks). Altogether, performance was obtained for 14 auditory temporal-processing measures. Young (N=20) and older (N=50) adults participated. Preliminary analyses (based on data from 50 of the 70 subjects) indicate that young listeners performed significantly better and with less variability than elderly listeners on all tasks. For most tasks, there was considerable overlap between the data from young and elderly listeners, indicating a modest negative impact of aging. At the individual level, correlational analyses among the older adults indicated that pure-tone thresholds were not predictive of temporal-processing performance and that performance on many of the temporal-processing tasks was moderately correlated. [Supported by NIA R01 AG022334.]

4pPPc8. MEG measures of the auditory steady-state response: Sinusoidal and non-sinusoidal stimuli. Garreth Prendergast (The University of York, Heslington, YO10 5DD York, UK, garreth.prendergast@ynic.york.ac.uk), Sam R. Johnson (The University of York, Heslington, YO10 5DD York, UK, sam@ynic.york.ac.uk), Gary G. Green (The University of York, Heslington, YO10 5DD York, UK, gary.green@ynic.york.ac.uk)

Human sensitivity to amplitude modulation has long been of interest to researchers, both in behavioural and neurological measures. Processing of amplitude modulation is implicated in the process of speech perception and sinusoidal amplitude modulation is used extensively to probe the mechanisms involved in encoding this information. The temporal envelope of speech is more accurately described as bursts of modulation rather than continuous modulation and the current work exposes participants to a continuum of modulation waveforms; from sinusoidal to pulsatile. Waveforms were amplitude modulated at 4 Hz and imposed upon a 500 Hz pure-tone carrier. The waveforms were generated using raised-cosine pulses with different half-durations. Half-durations of 8, 16, 24, 32, 64 and 125 ms were used (125 ms producing sinusoidal amplitude modulation at 4 Hz). Stimuli were 240 seconds in duration and responses were collected on a 248 channel whole-head MEG scanner. The frequency domain steady-state response was analysed from each condition in 14 participants, and results confirmed that the response to sinusoidal amplitude modulation was significantly lower than to modulations more representative of those found in speech signals. This suggests that non-sinusoidal stimuli may be more effective when investigating these auditory mechanisms.

4pPPc9. Laboratory synthesis of industrial noise environments with predetermined statistical properties. Wei Qiu (State University of New York, 101 Broad Street, Plattsburgh, NY 12901, USA, wei.qiu@plattsburgh.edu), Bob Davis (State University of New York, 101 Broad Street, Plattsburgh, NY 12901, USA, davisri@plattsburgh.edu), Roger P. Hamernik (State University of New York, 101 Broad Street, Plattsburgh, NY 12901, USA, roger.hamernik@plattsburgh.edu)

High-level nonGaussion noise is commonly found in a variety of industrial environments. Recent experiments have shown that for a given energy level, the statistical properties of a noise can have a strong effect on the extent of hearing loss produced in exposed individuals. In order to study, in an

animal model, the effects on hearing of such noise environments, the statistical properties of the noise as embodied in the kurtosis metric must be under experimental control. For a fixed value of kurtosis and energy level the following four variables will have a strong effect on hearing loss: (1) peak histogram; (2) interval histogram; (3) duration of noise transients; and (4) level of any background Gaussian noise. Simulations have shown that the relations among kurtosis and these variables are nonlinear. However, under certain restricted conditions, these relations may be linear. Accordingly, two strategies for designing controlled industrial noise exposures are presented: (1) the interval-priority model and (2) the duration-priority model. Computer simulations and measurements of actual acoustic environments showed that these two models could be effectively used to simulate a wide variety of realistic industrial noises.

4pPPc10. Intervention for restricted dynamic range and reduced sound tolerance. Charles Formby (University of Alabama, 700 University Boulevard East Suite 315, Tuscaloosa, AL 35487, USA, cformby@as.ua.edu), Monica Hawley (University of Maryland, 16 S. Eutaw St, Suite 500, Baltimore, MD 21201, USA, moncia@hawleyonline.net), Laguinn Sherlock (University of Maryland, 16 S. Eutaw St, Suite 500, Baltimore, MD 21201, USA, gsherlock@smail.umaryland.edu), Susan Gold (University of Maryland, 16 S. Eutaw St, Suite 500, Baltimore, MD 21201, USA, sgold@smail.umaryland.edu), Allyson Segar (University of Maryland, 0100 Lefrak Hall, College Park, MD 20742, USA, asegar@hesp.umd.edu), Christine Gmitter (University of Maryland, 0100 Lefrak Hall, College Park, MD 20742, USA, cgmitter@hesp.umd.edu), Justine Cannavo (University of Maryland, 0100 Lefrak Hall, College Park, MD 20742, USA, jcannavo@hesp.umd.edu)

Hyperacusis is an abnormal condition of sound intolerance that may cause some persons to reject amplified sound from their hearing aids. A significant secondary benefit reported for many patients receiving Tinnitus Retraining Therapy (TRT) is increased Loudness Discomfort Levels (LDLs). TRT involves both counseling and sound therapy (i.e., daily exposure to soft sound from bilateral noise generators (NGs)). We implemented a randomized, double-blind, placebo-controlled clinical trial to assess the efficacy of TRT as an intervention to improve sound tolerance in hearing-aid eligible persons with hyperacusis and/or restricted dynamic ranges. Subjects were assigned to one of four treatment groups: 1) full treatment, both counseling and NGs, 2) counseling and placebo NGs, 3) NGs without counseling, and 4) placebo NGs without counseling. They were evaluated at least monthly, typically for five months or more, on a variety of audiometric tests, including LDLs, the Contour Test for Loudness, and word recognition measured at comfortable and loud levels. Over 80% of the subjects assigned to full treatment achieved significant benefit (defined as shifts of greater than 10 dB in LDLs or the Contour Test uncomfortable level); whereas, most subjects assigned to a partial treatment group did not benefit from their treatment. [Supported by NIH].

4pPPc11. Relationship between a visual stimulus with a feeling of depth and its equivalent sound pressure level (ESPL). Hiroshi Hasegawa (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, hasegawa@is.utsunomiya-u.ac.jp), Hirotaka Ono (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, hasegawa@is.utsunomiya-u.ac.jp), Takumi Ito (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, hasegawa@is.utsunomiya-u.ac.jp), Ichiro Yuyama (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, yuyama@is.utsunomiya-u.ac.jp), Masao Kasuga (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, kasuga@is.utsunomiya-u.ac.jp), Miyoshi Ayama (Utsunomiya Univ., 7-1-2 Yoto, Tochigi-ken, 321-8585 Utsunomiya-shi, Japan, nakatsue@is.utsunomiya-u.ac.jp)

This study investigated the equivalent perception between a visual stimulus and its associated sound. Experiments of an auditory-visual stimulus presentation using an audio-video clip of a man beating a drum on a road were performed. The visual stimulus had a feeling of depth with a perspective view of the road. The visual stimulus was projected onto a screen that had the viewing angles of 43.8 deg.(W) * 25.4 deg.(H). Four kinds of distance between the subject and the visual stimulus from 5 to 40 m, seven kinds of the delay time between auditory and visual stimulus from -8 F to 8 F (1 F = 1/30 s), where ″+″ indicates that the visual event preceded the sound, and nine levels of the sound stimulus from -12 dB to 12 dB of the standard sound pressure level (SPL) were combined and presented. We evaluated the sound pressure level matching with each presentation pattern (equivalent sound pressure level; ESPL). As a result, we obtained that the ESPL tended to decrease when the delay time increased (the sound was delayed). This result shows a possibility that the visual stimulus was a little shifted to the direction of the sound.

4pPPc12. Temporal dynamics of stimulus specific processing in the human auditory cortex as revealed by electroencephalography. Paul M. Briley (MRC Institute of Hearing Research, University Park, NG7 2RD Nottingham, UK, paul@ihr.mrc.ac.uk), Katrin Krumbholz (MRC Institute of Hearing Research, University Park, NG7 2RD Nottingham, UK, katrin@ihr.mrc.ac.uk)

When the same sound is presented repeatedly, the electrical brain response recorded over the scalp decreases in amplitude, an effect known as adaptation. Adaptation is dependent on both the similarity of the sounds and the time between them. It has been particularly well studied for a deflection of the electrical response known as the N100, which peaks about 100 ms after sound onset and receives major contributions from auditory cortical sources. Adaptation may reflect decreased sensitivity to repetitive stimuli, but could also indicate more efficient processing of familiar events. Research on adaptation has often employed an alternating tone paradigm (A-B-A-B), examining the effects of changing inter-stimulus interval (ISI) or the frequency separation between A and B tones. Decreasing the frequency separation leads to an increase in N100 adaptation, and it has been suggested that the frequency specificity of this adaptation sharpens with decreasing ISI. In contrast, some studies have used A-B pairs with long inter-pair gaps and have found an enhancement of the N100 response to the B tone at short ISIs. In order to gain a better understanding of the processes contributing to adaptation and enhancement, this study investigates the temporal dynamics and the frequency selectivity of these effects.

4pPPc13. An Investigation of Width and Depth Perception toward a Sound Image Constructed of Multiple Variant Sound Waves Emitted from a Loudspeaker Array. Yoko Yamakata (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, yamakata@nict.go.jp), Toshiyuki Kimura (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, t-kimura@nict.go.jp), Munenori Naoe (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, nm.s512.ex@nict.go.jp), Michiaki Katsumoto (National Institute of Information and Communications Technology, 4-2-1, Nukui-Kitamachi, Koganei, 184-8795 Tokyo, Japan, katumoto@nict.go.jp)

Many musical instruments, including violins and guitars, vibrate their resonant bodies differently over their surface when they make a sound. This paper aims to reveal the influences of such vibration variation of a soundboard surface on the width and depth perception of the sound image when listeners were in a near-field 50 cm or 1 m away from the soundboard. In this paper, a loudspeaker array mimicked the surface vibration as each loudspeaker makes a corresponding sound independently and cooperatively. Three types of sounds, synthesized single-tone, multi-tone, and instrumental, were used as sources. To know what factors affect the perception of the sound image, various test sound sets were prepared by varying an original sound set in amplitude or delay for each frequency for each loudspeaker. Eight subjects were asked to identify which sound image in a pair of test sounds was wider or farther than the other according to Scheffe's pair comparison method. The results shows that a test sound set with a delay varia-

tion, which mimics sounds emitted by bending vibrations propagating on a soundboard, obviously influences the perception of sound image width and that the amplitude variation does not have much influence.

4pPPc14. MEG Recordings of Amplitude-modulated Noise and Tonal Stimuli in Healthy Adult Listeners. Yang Zhang (University of Minnesota, Dept. of Speech-Language-Hearing Sci. & Center for Neurobehavioral Development, Minneapolis, MN 55455, USA, zhang470@umn.edu), Yingjiu Nie (University of Minnesota, Dept. of Speech-Language-Hearing Sci. & Center for Neurobehavioral Development, Minneapolis, MN 55455, USA, niex0008@umn.edu), Toshiaki Imada (University of Washington, Dept. of Speech & Hearing Sciences, and Institute for Learning & Brain Sciences, Box 357988, Seattle, WA 98195, USA, imada@u.washington.edu), Keita Tanaka (Tokyo Denki University, Research Center for Advanced Technologies, 270-1382 Inzai, Japan, ktanaka@rcat.dendai.ac.jp), Masaki Kawakatsu (Tokyo Denki University, School of Information Environment, 270-1382 Inzai, Japan, kawakatu@asrl.dendai.ac.jp)

Amplitude modulation (AM) provides very important auditory information for the perception of complex sounds by normal listeners as well as cochlear implant users. The present study used a 122-channel whole-head magnetoencephalography (MEG) system to record auditory responses to amplitude-modulated pure tones and broadband noises in six healthy male adult subjects. The stimuli were presented in blocks of twenty with a brief silence in between, and the AM rates for both types of stimuli were at 20, 40, and 80 Hz. At least 80 artifact-free trials were collected for each stimulus. As expected, the MEG data showed a significant bilateral effect of AM rate in the N1m component. There was also strong evidence that the neural representations of both the unmodulated pure tone and noise stimuli in the auditory regions of both hemispheres could be significantly affected by the global context of block design stimulus presentation.

4pPPc15. Critical-band compression method of speech enhancement for elderly people: Investigation of syllable and word intelligibility. Keiichi Yasu (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, k-yasu@sophia.ac.jp), Hideki Ishida (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, ishida-h@sophia.ac.jp), Ryosuke Takahashi (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, ryo.t.sust@gmail.com), Takayuki Arai (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, arai@sophia.ac.jp), Kei Kobayashi (Dept. of Electrical and Electronics Engineering, Sophia University, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, kei-koba@ba2.so-net.ne.jp), Mitsuko Shindo (Sophia Univ. Research Center for Communication Disorders, 7-1 Kiyoi-cho, Chiyoda-ku, 102-8554 Tokyo, Japan, shindo-m@sophia.ac.jp)

Auditory filters for the hearing impaired tend to be wider than those of normal hearing people. Thus, the frequency selectivity decreases because of increased masking effects [Glasberg and Moore, J. Acoust. Soc. Am., 79(4), 1020-1033, 1986]. We have developed a method, called "critical-band compression," in which the critical band is compressed along the frequency axis [Yasu et al., Handbook of the International Hearing Aid Research Conference (IHCON), 55, Lake Tahoe, 2004]. We investigated whether our method improves syllable and word intelligibility. Thirty one elderly people participated in experiments. First, we measured the auditory filter bandwidth using a notched noise method [Patterson, J. Acoust. Soc. Am., 59(3), 640-654, 1976]. Second, we conducted syllable and word intelligibility tests. The compression rates of critical-band compression were set to 0% for the original, and 25%, 50%, and 75%. The results were that the percentages of correct responses were almost the same at 0%, 25% and 50% compression rates for syllable and word intelligibility. A significant correlation was not obtained between the compression rate of processing and the auditory filter bandwidth. [Work supported by JSPS.KAKENHI (16203041) and Sophia University Open Research Center from MEXT.]

4pPPc16. Speaker size discrimination for acoustically scaled versions of whispered words. Yoshie Aoki (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, 640-8510 Wakayama, Japan, s085065@sys.wakayama-u.ac.jp), Toshio Irino (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, 640-8510 Wakayama, Japan, irino@sys.wakayama-u.ac.jp), Hideki Kawahara (Faculty of Systems Engineering, Wakayama University, 930 Sakaedani, 640-8510 Wakayama, Japan, kawahara@sys.wakayama-u.ac.jp), Roy D. Patterson (Centre for the Neural Basis of Hearing, Department of Physiology, Development and Neuroscience, University of Cambridge, Downing Site, CB23EG Cambridge, UK, rdp1@cam.ac.uk)

Humans can extract the message from the voices of men, women, and children without being confused by the size information, and they can extract the size information without being confused by the message. This suggests that the auditory system can extract and separate information about vocal tract shape from information about vocal tract length. Smith et al. [J. Acoust. Soc. Am. 117(1), 305-318 (2005)], Ives et al. [J. Acoust. Soc. Am. 118(6), 3816-3822 (2005)], and Aoki et al. [ARO, 31st Midwinter meeting (2008)] performed discrimination experiments with acoustically scaled vowels, syllables, and naturally spoken words, respectively, and demonstrated that the ability to discriminate speaker size extends beyond the normal range of speaker sizes. Smith and Patterson [BSA Cardiff (2005)] demonstrated that performance on the size-discrimination task is only marginally reduced when the vowels are unvoiced. We extended these size-discrimination experiments to whispered versions of naturally spoken, four-mora Japanese words. The just-noticeable-difference for the whispered words was about 6 %, which is roughly the same as that for voiced words. The results show that voicing is not required for effective extraction of the size information. Research supported by JSPS Grant-in-Aid [B18300060] and the UK-MRC [G0500221].

4pPPc17. Effects of modality-dependent cuing and eye movements on sound localization. Beáta Tomoriová (Laboratory of Perception and Cognition, Technical University of Košice, Letná 9, 042 00 Košice, Slovakia, beata.tomoriova@gmail.com), Rudolf Andoga (Laboratory of Perception and Cognition, Technical University of Košice, Letná 9, 042 00 Košice, Slovakia, rudolf.andoga@tuke.sk), Norbert Kopčo (Laboratory of Perception and Cognition, Technical University of Košice, Letná 9, 042 00 Košice, Slovakia, kopco@tuke.sk)

A previous study of visual and auditory hemispheric cuing in horizontal sound localization found modality-dependent effects of cuing resulting in biases in responses [Kopco, Tomoriova, Andoga, J. Acoust. Soc. Am. 121, 3094, 2007]. The previous study also suggested that some of the effects might be due to eye movements as eye fixation was not controlled. The goal of the current study was to isolate the attentional effects from the eye movement effects. An experiment identical to the previous one was performed, with the exception that the subjects were fixating the center of the audiovisual display. Localization performance was measured for transient auditory stimuli originating in the frontal horizontal plane. In most runs, a cue preceded the stimulus and indicated (correctly or incorrectly) the hemisphere (left vs. right) from which the subsequent target arrived. The cues differed by modality and the cue-to-target onset asynchrony. The listeners were instructed to focus their attention to the cued side. Compared to the previous study, a reduction in some effects was observed. However, modality-dependent biases in performance persisted, confirming that auditory spatial attentional control is modality dependent and operating on time scale of seconds. [Supported by the Slovak Science Grant Agency.]