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Session 5aAA

Architectural Acoustics, Speech Communication, and Psychological and Physiological Acoustics: Psychological Aspects of Speech in Rooms I

John S. Bradley, Cochair
National Research Council, Inst. for Research in Construction, Ottawa, Ontario K1A 0R6, Canada

Hiroshi Sato, Cochair
National Inst. for Advanced Industrial Science and Technology, Inst. for Human Science and Biomedical Engineering, 1-1-1 Higashi, Tsukuba, Ibaraki 305-8566 Japan

Chair's Introduction—7:30

Invited Papers

7:35

5aAA1. Why we should design for optimum reverberation times in rooms for speech communication. J. S. Bradley (Inst. for Res. in Construction, Natl. Res. Council, Montreal Rd., Ottawa, Canada K1A 0R6)

Various texts list recommendations for optimum reverberation times as ideal goals in rooms for speech communication. However, some newer documents, such as ANSI S12.60, talk about maximum recommended reverberation times in rooms for speech communication. The source of the changed approach can be traced to interpretations of experimental results that do not consider the full implications for speech communication in real rooms, and also to the fact that reverberation time is not an ideal predictor of the quality of speech communication. In the extreme, minimizing reverberation times would lead to near anechoic rooms for speech and inadequate signal-to-noise ratios. The need for optimum reverberation times can be explained as a simple need to first achieve adequate signal-to-noise ratios. However, a more complete understanding is obtained by examining the benefits of early-arriving reflections of speech sounds on the intelligibility of the speech to listeners. Attempts to determine optimum reverberation times for normal hearing listeners, which are based on a balance between avoiding excessive reverberation and maintaining adequate signal-to-noise ratios, lead to a range of acceptable values that can vary with the ambient noise level. This discussion will be supported with examples from room acoustics measurements in classrooms and meeting rooms.

7:55

5aAA2. The relation between speech transmission index, clarity, and reverberation time and listening difficulty in the impulse response database of AIJ. Hiroshi Sato (Natl. Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-1-1 Higashi, Tsukuba, Ibaraki 305-8566, Japan), Yoshio Nishikawa (Konoike Co., Ltd., Tsukuba 305-0003, Japan), Hayato Sato, and Masayuki Morimoto (Kobe Univ., Nada, Kobe 657-8501, Japan)

The Speech Communication Research Working Group of AIJ (Architectural Institute of Japan) is collecting information on rooms including digitized impulse responses (IRs) to establish a database for evaluating and designing the speech transmission quality of rooms. This database consists of 966 measured IRs. This study presents the relationships between speech transmission index (STI), clarity (Cx) and reverberation time (T) as measures to consider for the design and evaluation of the speech transmission performance of rooms. The data show a wide range of STI and Cx values for a given T, and the minimum STI at each reverberation time can be obtained by diffused field theory. STI and Cx are seen to be better indicators than T for the design of rooms for speech and that T is not as good. Relationships between STI, Cx, and listening difficulty ratings from previous studies [Proc. of RADS (2004), Proc. of Forum Acusticum, pp. 1713–1718 (2005)] found both STI and Cx can be used as predictors of listening difficulty ratings. Finally, listening difficulty ratings of individual IR in the database are estimated and the distribution of listening difficulty ratings in a variety of rooms are presented as a solid bases to design speech transmission quality of rooms.

8:15

5aAA3. Unifying approaches for modeling and predicting speech intelligibility. Adelbert W. Bronkhorst and Sander J. van Wijngaarden (TNO Human Factors, Kampweg 5, 3769 DE Soesterberg, The Netherlands, adelbert.bronkhorst@tno.nl)

Speech intelligibility is practically always affected by the acoustic environment (reverberation, interfering speech, noise). When speech is processed electronically, distortions introduced by the transmission chain can also deteriorate intelligibility. Furthermore, intelligibility can be strongly influenced by nonacoustic factors such as contextual information, non-nativeness, and hearing impairment. Various intelligibility prediction methods have been developed (including SII, STI, SRS, and PESQ), which cover almost all factors influencing intelligibility. However, each individual method has limited applicability and no attempts have been made to unify them. Recently, the application domains of two widely used methods, the SII and the STI, have been significantly extended. They now, for example, cover interference that fluctuates in level, live speech, and digital transmission channels. These methods are actually very

5a SAT. AM

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similar in the way in which they treat frequency-domain effects. Because of this, it is not difficult to fuse the STI and SII into a single method with an unparalleled application domain. Still, important factors such as informational masking and context effects cannot be addressed and more effort should be invested in combining the SII/STI with other, less similar methods (such as the SRS) in order to develop a single, powerful, standard model for predicting speech intelligibility.

8:35

5aAA4. Influence of word familiarity on spoken word recognition. Shigeaki Amano, Tadahisa Kondo (NTT Commun. Sci. Labs., NTT Corp., 2-4 Hikari-dai, Seika-cho, Souraku-gun, Kyoto 6190237, Japan, amano@cslab.kecl.ntt.co.jp), Shuichi Sakamoto, and Yôiti Suzuki (Tohoku Univ., Japan)

Spoken word recognition is affected by many factors including sound pressure level, signal-to-noise ratio, word familiarity, word frequency, and phoneme sequence plausibility. Of these factors, word familiarity is assumed to have a strong effect but has received little attention. One reason for this is that word familiarity data for most words have been unavailable. However, we can now confirm the word familiarity effect because in 1999 Amano and Kondo published a word familiarity database for about 80 000 Japanese words. Experimental results clearly show that spoken words with a high familiarity are recognized better and faster than those with a low familiarity. In addition, word familiarity correlates with the recognition score of spoken words much better than word frequency, which was previously regarded as an influential factor in word recognition. Because the word familiarity effect is so strong, it is very important to control the familiarity of spoken words to obtain stable and reasonable results when assessing the speech hearing ability of individuals, and when measuring speech clarity in communication media and public spaces.

8:55

5aAA5. Development of word lists for word intelligibility tests in Japanese. Shuichi Sakamoto, Yôiti Suzuki (Res. Inst. of Elec. Commun., Grad. School of Info. Sci., Tohoku Univ., 2-1-1, Katahira, Aoba-ku, Sendai, 9808577, Japan, saka@ais.riec.tohoku.ac.jp), Shigeaki Amano, and Tadahisa Kondo (NTT Corp., Seika-cho, Soraku-gun, Kyoto, 6190237, Japan)

Speech intelligibility tests are widely used to evaluate personal speech hearing ability accurately. Although various lists have been proposed for the tests, almost all lists in Japan do not properly consider word difficulty as an aspect of speech recognition. Consequently, even if these tests are performed under identical conditions, correct results cannot be obtained. We have been developing word lists for word intelligibility tests to cope with these problems. We control word familiarity, which strongly affects word recognition scores, to equalize difficulty in recognition between word lists. Words are divided into four word-familiarity ranks. Twenty lists of 50 words are constructed in each rank by considering phonetic balance. Moreover, to compensate slight differences in intelligibility scores among word lists within the same word-familiarity rank, the relationship between the difference of word familiarity and signal-to-noise ratio (SNR) is estimated. Then this relationship is used to compensate the difference of the intelligibility score. The other compensation method is based on speech recognition threshold (SRT). The intelligibility score of each word can be equalized by changing the SNR according to the difference of the SRT of the word. The effectiveness of these compensation methods is discussed.

9:15

5aAA6. Binaural and spatial factors affecting speech intelligibility in rooms. H. Steven Colburn, Suzanne Carr, and Rui Wan (Hearing Res. Ctr. and Dept. of Biomed. Eng., Boston Univ., 44 Cummington St., Boston, MA 02215, colburn@bu.edu)

Prominent complaints of hearing-impaired listeners relate to the difficulties they experience in complex acoustic environments, particularly environments with multiple sources and reverberation. Available data and some theoretical ideas about the nature of the difficulties with speech intelligibility in these environments will be reviewed with impaired listeners in mind. The effects of the spatial characteristics of the sound sources and the room on the binaural signals and their processing is the primary focus of this presentation. Listeners with normal hearing as well as listeners with hearing impairments, with and without hearing aids, will be considered. Theoretical topics to be discussed include the effects of reverberation on interaural differences and on temporal modulation patterns, including the consequences for performance as predicted by standard models of binaural processing. [Work supported by NIH R01 DC00100.]

9:35

5aAA7. The effect of noise on novel word learning in sequential bilingual children. Pui Fong Kan, Kathryn Kohnert, and Peggy Nelson (Dept. of Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455)

Preschool children must learn in noisy environments (Picard and Bradley, 2001), but young children are more negatively affected by background noise than are adults (Elliott, 1979). Young second-language learners are more negatively affected by noise than are monolingual children (Crandell *et al.*, 1996; Nelson *et al.*, 2005). Background noise may be still more detrimental in the case of preschool children who speak one language (L1) at home and who start to learn a second language (L2) in nursery school. In this present study we examine the effect of noise on fast-mapping skills in L1 (Hmong) and L2 (English). In the fast-mapping task each child was briefly presented with 5 novel wordscounterbalanced in Hmong and in English−in a quiet setting over two different sessions. Another 5 novel Hmong and English words were presented to each child in two sessions in the presence of babble background noise. During each session the children were tested immediately after their exposure to the novel words. The results confirmed (1) that the children demonstrated−in both L1 and L2−better skills in quiet than in noise and (2) that the childrens fast-mapping skills were positively correlated with their pre-existing L1 and L2 language skills.

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9:55–10:15 Break

10:15

5aAA8. The effects of fluctuating interaural cues on the segregation of speech in rooms. Douglas Brungart and Nandini Iyer (Air Force Res. Lab., 2610 Seventh St., Bldg 441, WPAFB, OH 45433, douglas.brungart@wpafb.af.mil)

Spatial separation is known to improve the segregation of talkers in anechoic environments, but relatively little is known about the role spatial cues play in speech segregation in reverberant rooms. One might expect the random disruptions in the interaural time and level differences (ILDs and ITDs) that occur in reverberant environments to eliminate many of the intelligibility benefits that normally occur for spatially separated speech. However, the precedence effect often leads to a robust perception of spatial location even in extremely echoic environments. This can result in an apparent separation between talkers that may facilitate selective attention to the location of the target speech even in a highly reverberant room. Also, random fluctuations in ITD and ILD may lead to differences in the apparent source widths of the target and masking sounds, which might serve as a segregation cue. In this talk, we examine the effects fluctuating interaural difference cues have on the segregation of target speech from competing speech or noise. The results suggest that differences in apparent source width can be used to segregate competing speech signals even when the target and masking signals appear to originate from the same direction relative to the listener.

10:35

5aAA9. The intelligibility advantage of clear speech in noise and reverberation backgrounds: Effects of speaking rate. Jean C. Krause (Commun. Sci. and Disord. Dept., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33617, jkrause@cas.usf.edu)

Clear speech refers to a speaking style that is significantly more intelligible than conversational speech for a variety of listeners and backgrounds, including roomlike backgrounds such as noise and reverberation [Payton *et al.*, J. Acoust. Soc. Am. **95**, 1581–1592 (1994)]. Although typically spoken more slowly than conversational speech [e.g., Uchanski *et al.*, J. Speech Hear. Res. **39**, 494–509 (1996)], talkers can produce clear speech at normal rates with training [Krause and Braida, J. Acoust. Soc. Am. **112**, 2165–2172 (2002)]. Whether this form of clear speech (clear/normal speech) provides similar intelligibility benefits to clear (clear/slow) speech in all situations has not been fully characterized, though some results [Krause and Braida, Iranian Audiol. **2**, 39–47 (2003)] suggest that the amount of benefit is more dependent on talker and environment. To investigate this possibility for roomlike degradations, the intelligibility advantage provided by clear/slow and clear/normal speech was evaluated by groups of eight normal-hearing listeners at five signal-to-noise ratios (-4, -2, 0, $+2$, and $+4$ dB) and three levels of reverberation (0, 0.18, and 0.6 s). Results for four talkers will be reviewed in order to illustrate the effects of speaking rate and speaking style on speech intelligibility in rooms.

10:55

5aAA10. Separating and understanding a talker from a mixture in reverberant spaces. Barbara G. Shinn-Cunningham, Madhusudana Shashanka, and Scott Bressler (677 Beacon St., Boston, MA 02215)

Many studies show that reverberant energy can interfere with the ability to understand a listener by smearing out the temporal modulations and distorting the spectral content that convey speech meaning. However, the amount of distortion caused by the reverberation in everyday settings often is not severe enough to degrade speech understanding when there is only one talker in a room. Despite this, even modest reverberation can have an enormous impact on the ability to understand the same talker when there is another sound source in the environment. One factor that likely contributes to the problem of understanding a talker in a mixture of sounds in reverberant spaces is that it is difficult to separate the target talker from the competing sounds. This talk will explore the ways in which reverberant energy degrades the ability to separate sounds in a mixture. The effects of reverberation on different cues for source segregation, including pitch and location, will be discussed. [Work supported by AFOSR and NIH.]

11:15

5aAA11. Preprocessing speech against reverberation. Takayuki Arai (Dept. of Elec. and Electron. Eng., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo, 102-8554 Japan)

Although short reverberation in a room may help speech perception, it is known that long reverberation causes degradation in speech comprehension. This is especially true for elderly people, the hearing-impaired, and non-native listeners. In order to prevent intelligibility degradation, we can apply a signal processing technique to the public address system of a room before speech signals are radiated through the loudspeakers. So far, we have developed several such preprocessing techniques. The two main techniques are modulation filtering [Kusumoto *et al.*, Speech Commun. **45**, 101–113 (2005)] and steady-state suppression [Arai *et al.*, Acoust. Sci. Technol. **23**, 229–232 (2002)]. Both techniques essentially enhance the temporal dynamics of speech between 1 and 16 Hz [Arai *et al.*, J. Acoust. Soc. Am. **105**, 2783–2791 (1999)]. Steady-state suppression, which suppresses steady-state portions of speech, reduces overlap-masking and improves speech intelligibility for young, elderly, and non-native listeners in reverberant environments [e.g., Hodoshima *et al.*, J. Acoust. Soc. Am. **119**, 4055–4064 (2006)]. This technique is more effective when speech-rate slowing is applied in advance [Arai *et al.*, Acoust. Sci. Tech. **26**, 459–461 (2005)]. Intelligibility with this technique exceeds a simple speech-rate slowing approach. [Work partially supported by JSPS.KAKENHI (16203041).]

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11:35

5aAA12. Failure of the precedence effect in children and in persons with hearing loss. Ruth Litovsky (Univ. of Wisconsin Waisman Ctr., 1500 Highland Ave., Madison, WI 53705)

Children spend a vast majority of time in reverberant environments. However, their auditory system appears not to be as fully equipped as that of adults for coping with reverberation. In our studies we simulate simple aspects of reverberation by exploiting a well-known auditory illusion known as the precedence effect (PE). Stimuli consist of clicks or speech, pairs of source (lead), and its first-arriving reflection (lag). Results from typically developing normal-hearing children ages 3–10 suggest that the PE undergoes significant maturation during childhood. In addition, we studied 10-year-old children with specific language impairment (SLI), who are delayed on expressive and receptive language. Performance was similar to that of younger, language-age-matched children, suggesting that the PE could be a useful clinical tool for measuring nonverbal temporal information processing. The PE was also used to study deaf adults with bilateral cochlear implants. Results suggest that binaural sensitivity is only available when the cochlea can be directly stimulated, but in free field the temporal smearing caused by the microphone leads to degraded and abnormal PE. Taken together, these studies suggest that a basic and necessary auditory function, that of suppressing echoes, fails under numerous circumstances, in children and in deaf, bilateral cochlear implant users.

SATURDAY MORNING, 2 DECEMBER 2006 LANAI ROOM, 8:00 A.M. TO 12:00 NOON

Session 5aAB

Animal Bioacoustics: Animal Bioacoustics Poster Session (Poster Session)

Sheryl L. Coombs, Cochair
Bowling Green State Univ., Dept. of BioScience, Life Science Bldg., Bowling Green, OH 43403

Yutaka Hosokawa, Cochair
Ryukyu Univ., Dept. of Physiology, 207 Uehara, Nishihara-chu, Okinawa 903-0215, Japan

Contributed Papers

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

5aAB1. Reconstruction of bat sonar beams using a multichannel microphone array. Peter Stilz, Wiebke Pflästerer, and Hans-Ulrich Schnitzler (Tierphysiologie, Universität Tübingen, Auf der Morgenstelle 28, 72076 Tübingen, Germany, peter.stilz@uni-tuebingen.de)

We reconstruct the sonar beams emitted subsequently by flying bats from recordings of a two-dimensional microphone array. Based on the bats' position in space, which can be accessed through a infrared 3D-video system, and the simultaneous recordings of 16 small microphones, the sonar beams emitted by the bats and the sonar footprints on arbitrary planes can be estimated. For the interpolation of a continuous sonar beam at different frequency bands, directional frequency responses, atmospheric and geometric attenuation, and different lag times are taken into account for each microphone. Mean and maximum call intensities can be analyzed and various analysis parameters can be adjusted, different visualizations are available, and numerical data are exported. The system provides access to estimations of the maximum intensity of each call in front of the bat, beam width, shape, and intensity for arbitrary frequency bands, the aiming of the beam, and the bat's scanning behavior on its flight path.

5aAB2. A new low-power acoustic bat detector for long-duration observations. Matt Heavner (Univ. of Alaska Southeast, 11120 Glacier Hwy., Juneau, AK 99801)

A new small, low-power, long-duration, field-deployable computer for the acoustic monitoring of bats in Southeast Alaska has been developed. Parker *et al.*, (1997) report the northern geographic limit of four bat species to be Southeast Alaska. (A fifth species, *E. fuscus*, goes much farther north.) The relationship of the bats to forest management practices, the habitat usage of the bats, and population size and trends are all very poorly known for bats in Southeast Alaska. Long duration monitoring of several different types of area (such as old-growth versus recently logged forest) will provide knowledge to improve management practices in regards to bat ecology in Southeast Alaska. With the motivations just described, the hardware, development methods, and analysis software designed to develop an improved detector are presented. Tradeoffs between computation power, power consumption, cost, and other factors will be discussed. The project has benefitted in several direct ways from the system design requirement of modularity. [Work supported by Alaska epartment of Fish & Game.]

5aAB3. Development of networked weatherproof remote sensing embedded Linux system for bioacoustical evaluation. Hiroki Kobayashi, Yoshihiro Kawasaki, and Shinichi Watanabe (Matsushima Lab., Dept. of Geosystem Eng., The Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo, 113-8656, Japan, tt56958@mail.ecc.u-tokyo.ac.jp)

The purpose of this research is to develop a real-time evaluation technology of bioacoustic diversity in a remote location by networked Linux embedded system. The hardware is designed to operate 24 h a day, 7 days a week, 365 days a year under severe environment, such as extremely high moisture and temperature environment in subtropical forest. Three stages of the hardware development were completed at this point. At first, a wired long-lived bioacoustic monitoring system in subtropical forest in Iriomote Island, Japan was developed and succeeded to operate continuously for the

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past 2 years based on an unmanned condition with less than 1000 US dollars annual budget. Based on this development, a wired multifunctional monitoring system with infrared cameras and microclimate monitoring sensors was developed and evaluated with less than 4000 US dollars. At last, with FOMA (Freedom of Mobile Multimedia Access) mobile information technology offered by Japanese mobile phone operator NTT DoCoMo, a multifunctional bioacoustic monitoring system with two-way communication system which operates passive and active observation for Iriomote Cat was developed with less than 8000 US dollars. [Work supported by NTT DoCoMo.]

5aAB4. Can birds discriminate between simple sounds and natural vocalization with different degrees of reverberation. Sandra H. Blumenrath and Robert J. Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742)

Birds in natural habitats using long-range acoustic signals experience substantial acoustic alterations of their signals when transmitted from a sender to a receiver. The main degradation comes from reflections of the emitted sounds from surfaces in and around the transmission pathway. These reverberations cause tails of echoes and affect frequency and amplitude modulation within the sounds' original time frame. Humans discriminate among reverberant and nonreverberant speech segments and use the nature of reverberation in determining the location of the sound source. Using operant conditioning and psychophysical methods, we investigated the ability of several species of small birds to discriminate between nondegraded sounds and sounds artificially reverberated to varying degrees, while holding other sound aspects constant. Birds were able to easily discriminate between nonreverberant sounds and sounds reverberated to various extents. By comparing the birds' performance on stimuli varying along different dimensions of reverberation, we can make predictions as to the birds ability to use reverberations as acoustic cues in distance estimation and locating sound sources. [Work supported by NICD DC000198 (to R.J.D.) and DC004664.]

5aAB5. Perception of budgerigar (*Melopsittacus undulatus*) warble elements. Hsiao-Wei Tu and Robert J. Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742)

The warble song of budgerigar (*Melopsittacus undulatus*) contains various classes of vocal elements, including contact calls, alarm calls, harmonic calls, etc. Previous work has shown that budgerigars can discriminate different call types according to both acoustic and functional features; however, there is no evidence on how well birds can distinguish subtle differences within and between each category. Using operant conditioning and psychophysical methods, budgerigars were trained to discriminate among different warble elements. In general, birds were able to discriminate quite easily between all warble elements, even those with high spectral cross correlations. Humans, on the other hand, had difficulty distinguishing among warble elements that were similar to one another. The relative discriminability of warble elements within and across categories is used to more precisely define the functional categories of warble elements in understanding vocal communication in this species. [Work supported by NIDCD DC000198 to R.J.D. and P30 DC004664.]

5aAB6. Voices of the dead: Complex nonlinear vocal signals from the larynx of an ultrasonic frog. Roderick A. Suthers (School of Medicine and Dept. of Biol., Indiana Univ., Bloomington, IN 47405, suthers@indiana.edu), Peter M. Narins (Univ. of California, Los Angeles, CA 90095), Wen-Yu Lin (Univ. of Illinois, Urbana, IL 61801), Hans-Ulrich Schnitzler, Annette Denzinger (Univ. Tuebingen, D-72076 Tuebingen, Germany), Chun-He Xu (Chinese Acad. of Sci., Shanghai, P. R. China), and Albert S. Feng (Univ. of Illinois, Urbana, IL 61801)

Most anurans are highly vocal but their vocalizations are stereotyped and simple with limited repertoire sizes compared to other vocal vertebrates, due presumably to the limited mechanisms for fine vocal motor control. We recently reported that the call of the concave-eared torrent frog *(Amolops tormotus)* is an exception in its seemingly endless variety, musical warbling quality, extension of call frequency into the ultrasonic range, and the prominence of nonlinear features such as period doubling. We now show that the major spectral features of its calls, responsible for this frog's vocal diversity, can be generated by forcing pressurized air through the larynx of euthanized males. Laryngeal specializations for ultrasound appear to include very thin portions of the medial vocal ligaments and the reverse sexual size dimorphism of the larynx being smaller in males than in females. The intricate morphology of the vocal cords, which changes along their length, suggests that nonlinear phenomena likely arise from complex nonlinear oscillatory regimes of separate elastically coupled masses. *Amolops* is thus the first amphibian for which the intrinsic nonlinear dynamics of its larynx, a relatively simple and expedient mechanism, can account for the species call complexity, without invoking sophisticated neuromuscular control.

5aAB7. Comparative analysis of head-banging behavior in the subterranean termites *Coptotermes formosanus* and *Reticulitermes flavipes* using high-speed imaging. Tom Fink, Lichuan Gui, Yong Wang, Zhonghua Cao (Natl. Ctr. for Physical Acoust., One Coliseum Dr., University, MS 38677), Adarsh Jaiswal, Orwa Tahaineh, Roger Hasse, John Seiner (Natl. Ctr. for Physical Acoust., University, MS 38677), and Alan Lax (U.S. Dept. Agriculture, Blvd. New Orleans, LA 70124)

Disturbed soldiers of *Coptotermes formosanus* (FST) readily headbang (HB) only in their galleries and carton nests while *Reticulitermes flavipes* (Rf) will also readily HB in an artificial chamber. The headbanging motion was examined using the Photron FASTCAM-ultima APX video camera at 10 000 frames per second. Termites were placed in a 0.25-in. diameter hole in a wood block and induced to HB by touching them with a small brush. A PCB Piezotronics 352B accelerometer was used to detect HB-induced vibrations. These vibration signals were then received by the Photron Multi-Channel-Data-Link apparatus, which then superimposes the vibration waveform on the video frames corresponding to the head movement. Thus, movement of the head can be correlated with the vibrations produced. Time between successive HB is about 70 ms or more in FST vs 40 ms or less in Rf despite the much larger head of Rf. Upward and downward head velocities in FST are, respectively, 100 and 240 mm/s, while the respective values for *Reticulitermes* are over 200 and about 400 mm/s. Acceleration for FST is about 400 m/s vs up to 700 m/s^2 for *Reticulitermes*. Both species may show head rebounding after the initial contact with the substratum.

5aAB8. Underwater auditory localization by harbor seals (*Phoca vitulina*). Anaïs Bodson, Lars Miersch, and Guido Dehnhardt (General Zoology & Neurobiology, Univ. of Bochum, ND 6/33, D-44780 Bochum, Germany, bodson@marine-science-center.de)

The underwater sound localization acuity of a swimming harbor seal was measured in the horizontal plane. The stimulus was either a double sound (two 6-kHz pure tones lasting 0.5 s separated by an interval of 0.2 s) or a single continuous sound of 1.2 s. Testing was conducted in a 10-m-diameter underwater half-circle with hidden loudspeakers. The animal was trained to swim along the diameter of the half-circle and to change its course towards the sound source right after the signal was given. The seal indicated the sound source by touching its assumed position at the half-circle. The deviation of the seal's choice from the actual sound source was measured by means of video analysis. In trials with the double sound the seal localized the sound sources with a mean deviation of 2.8° and in trials with the single sound with a mean deviation of 4.5°. In a second experiment minimum audible angles of the animal were determined as a function of frequency (2, 4, 6, 8, and 16 kHz resulting in MAAs of 9.4°, 9.6°, 9.8°, 13.0°, and 13.5°, respectively). Intraspecific differences of the MAA will be presented for five additional harbor seals.

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5aAB9. Evaluation of the auditory capabilities of marine mammals using a portable auditory-evoked potential system. Kristen A. Taylor, Paul E. Nachtigall, T. Aran Mooney, Michelle M. Yuen (Marine Mammal Res. Program, Hawaii Inst. of Marine Biol., Univ. of Hawaii, P.O. Box 1106, Kailua HI 96734), and Alexander Ya Supin ((Russian Acad. of Sci., Moscow, Russia)

There is growing concern about the increasing amounts of acoustic pollution in our oceans. Anthropogenic noise sources span a large range of frequencies and amplitudes. The extents to which these noise sources adversely impact marine mammals is poorly understood yet they are of particular significance given that the acoustic environment is crucial for many aspects of marine mammal life. Past and present experiments have addressed the effects of different sounds on the hearing capabilities of captive dolphins. However, only a few individuals from a small number of species have been tested. Our research group has created a portable system that is capable of measuring the hearing thresholds of marine mammals using auditory-evoked potential (AEP) techniques. This portable system has enabled us to obtain audiograms of otherwise inaccessible marine mammal species. This system will also allow us to measure the hearing of stranded animals as well as begin to quantify the normal variations in hearing between different ages and genders of animals in captivity.

5aAB10. Optical imaging of neural activity in the right and left guinea pig auditory cortices evoked by repeated sound. Yutaka Hosokawa (Dept. of Physiol., Ryukyu Univ., Nishihara-tyo, Okinawa, 903-0215 Japan) and Junsei Horikawa (Toyohashi Univ. of Technol., Toyohashi, 441-8580 Japan)

Processing of spectral and temporal information in the right and left auditory cortices of guinea pigs anesthetized with ketamine and xylazine was investigated using an optical imaging technique with a voltage-sensitive dye (RH795). Tonotopical organization in the primary (AI) and the dorsocaudal (DC) fields in the left and the right auditory cortices were compared, as visualized by superimposition of the areas activated by tones at 2, 4, 8, and 16 kHz. In 65% of 31 animals, the distance between 2- and 16-kHz isofrequency bands was longer in the left than in the right AI; it was shorter in the left than in the right DC. Repetition rate transfer functions (RRTFs) in AI measured with repeated sounds (420 Hz repetition rates) were low pass, with a sharp drop-off above 10 Hz. In contrast, RRTFs in DC were bandpass with a peak at 8 or 10 Hz. The cutoff frequencies of RRTF were different for each of the left and right cortices.

5aAB11. The physoclistous swim bladder of chaetodontid butterflyfishes: Implications for acoustic function. Christopher Woods, Jacqueline Webb (Dept. of Biol., Villanova Univ., Villanova, PA 19085; jacqueline_webb@mail.uri.edu), and Darlene Ketten (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Butterflyfishes (genus *Chaetodon*) have a swim bladder lateral line connection (laterophysic connection, LC), hypothesized to convert sound-induced oscillations of the swim bladder into fluid flow in the lateral line system and/or ear. Evaluation of LC function is dependent upon an understanding of swim bladder acoustics, which is a function of swim bladder anatomy. We used several anatomical methods (including CT) to describe swim bladder and tunica externa and tunica interna morphology (including gas gland/*rete mirabile* complex and the oval, responsible for gas secretion and resorption) in *Chaetodon* and *Forcipiger*. Swim bladder and tunica externa morphology differ between *Chaetodon* spp. with different LCs morphologies. A perforated transverse diaphragm divides the gas volume into two compartments. The gas gland/*rete mirabile* complex is in the ventral midline of the anterior compartment; it varies in morphology and is largest in species with a direct LC. The oval, defined by an extensive capillary network, occupies the entire tunica interna of the posterior compartment, so that diaphragm position determines oval size. These data raise questions about swim bladder structure-function relationships with respect to the reception and transduction of acoustic stimuli in coral reef fishes. [Work supported by NSF Grant IBN-0132607 to J.F.W.]

5aAB12. The temporal resolution of goldfish hearing: An auditory evoked potential study of gap detection. Jianqiang Xiao and Christopher Braun (Dept. of Psych., Hunter College, 695 Park Ave., New York, NY 10021, jxiao@hunter.cuny.edu)

Temporal processing in goldfish (*Carassius auratus*) was studied by measuring auditory evoked potentials (AEP) using gaps in continuous band-limited Gaussian noise. Long (>100 ms) silent gaps in high amplitude noise (30 dB SL) evoked distinct offset and onset responses, lasting 6090 ms. Offset and onset responses overlapped with shorter gaps and became difficult to distinguish for gaps <10 ms. The gap-response waveform was modeled as the sum of offset and temporally-shifted onset responses from the same animals (using 120-ms gaps). This model predicted the waveform of gap responses nearly perfectly for gaps longer than 6 ms. Waveforms evoked by shorter gaps (<6 ms) did not accurately fit this model, suggesting that one or both components of the gap response were inhibited or altered. Nonetheless, clear responses were evoked by gaps shorter than 1 ms at this intensity, while longer gaps (up to 10 ms) were required for detection in a low-intensity noise background. This study extends and confirms prior reports that the temporal resolution for goldfish is on the order of 1 ms, but there is also evidence for longer integration processes, over a period of several ms. [Work supported by Grants S06GM60654, R03MH067808, RR03037 (NIH).]

5aAB13. The effects of source distance and orientation on lateral line stimulation patterns and source localization by Lake Michigan mottled sculpin. Sheryl Coombs and Paul Patton (Dept. of Biological Sci., JP Scott Ctr. for Neurosci., Mind, and Behavior, Bowling Green State Univ., Bowling Green, OH 43403)

Visually deprived, Lake Michigan mottled sculpin respond to small dipole sources (sinusoidally vibrating spheres) with an initial orienting response and subsequent approach and strike behaviors. Lateral line canal neuromasts, but not superficial neuromasts, are required for the initial orienting response. Computational (potential flow) models reveal that stimulation patterns across lateral line canals vary in complex ways when the orientation (axis of vibration) and distance of the dipole source are varied. The body location where lateral line stimulation is maximal (the excitatory peak) might seem to provide a reliable cue of source location. However, this cue is (1) confounded by the relative amplitude, number, and body location of excitatory peaks when source orientation is varied and (2) degraded by both an increase in peak width and a reduction in peak amplitude when source distance is increased. Behavioral measures of orienting accuracy for equally detectable sources at two distances (3 and 6 cm) and horizontal orientations (parallel or orthogonal to the long axis of the fish) reveal that localization abilities are degraded by increases in source distance. Although varying source orientation does not produce dramatic deficits in orienting accuracy, more subtle effects were observed that may relate to information encoded in spatial excitation patterns.

5aAB14. The use of multimodal communication during courtship in Malawi cichlids. Adam R. Smith and Moira J. van Staaden (Dept. of Biological Sci. and J. P. Scott Ctr. for Neurosci., Mind & Behavior, Bowling Green State Univ., Bowling Green, OH 43403)

The cichlids of Lake Malawi are often considered the quintessential example of a species radiation driven primarily by sexual selection and

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relying on visual cues. However, the description of sound production by courting males in several species raises the possibility that acoustic signaling may play a significant role in the selection process. This study describes the mating calls from multiple species in the genera *Melanochromis* and *Metriaclima* in order to discern if there is active selection of call characteristics between closely related species or whether differences are more likely the result of drift. Variation in total call duration and fundamental frequency within genera often exceeded that between genera, particularly between the sister species *Melanochromis johanni* and *Melanochromis cyaneorhabdos*. This suggests that mating calls may indeed be subject to divergent sexual selection in closely related species. Moreover, *in situ* sound recordings from Lake Malawi indicate that this behavior is of ecological relevance and not simply an artifact of laboratory conditions. Together, these results suggest that current models of sexual selection in Malawian cichlids may benefit from consideration of previously ignored sensory modalities. The plasticity inherent in such multimodal processing may prove of particular importance to the species-rich African cichlid radiation.

5aAB15. Reaction of plaice (*Pleuronectes platessa*) and cod (*Gadus morhua*) to low-frequency sound in an experimental tank. Christina Mueller-Blenkle (FRS Marine Lab., P.O. Box 101, Aberdeen AB11 9DB, UK) and Karin Luedemann (Institut fr Angewandte Fischbiologie, 22083 Hamburg, Germany)

Anthropogenic sound in the sea is an increasing problem and more knowledge on the influence of sound on fish needs to be collected. For this reason experiments are carried out in an experimental tank 10 m in diameter to examine the reaction of juvenile and adult cod and plaice to sound. Pure tones from the frequency range of offshore wind turbines are artificially emitted in typical wind turbine sound levels of 130 and 140 dB *re* 1 Pa. A sound gradient in the tank enables the fish to avoid highest sound levels. The number of fish in the vicinity of the sound source is determined in regular time intervals before, during, and after the 24-h period of sound exposure. Changes in numbers of fish can be seen in the periods with and without sound in many experiments. Reactions of plaice are ambiguous, pointing to both avoidance of or attraction to the sound. In the majority of the experiments carried out on cod, avoidance behavior to the sound can be seen with strongest reactions at 25, 60, and 90 Hz. The results indicate potential avoidance behavior of cod to the operational noise of offshore wind farms.

SATURDAY MORNING, 2 DECEMBER 2006 KAHUKU ROOM, 9:00 TO 11:10 A.M.

Session 5aBB

Biomedical Ultrasound/Bioresponse to Vibration: Ultrasound Contrast Agents

Charles C. Church, Cochair

Univ. of Mississippi, National Center for Physical Acoustics, 1 Coliseum Dr., University, MS 38677

Chair's Introduction—9:00

Contributed Papers

9:05

5aBB1. Microbubble spectroscopy of ultrasound contrast agents. Sander van der Meer, Benjamin Dollet (Phys. of Fluids, Univ. of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands), Chien T. Chin, Ayache Bouakaz, Marco Voormolen, Nico de Jong (Erasmus MC, 3000 DR Rotterdam, The Netherlands), Michel Versluis, and Detlef Lohse (Univ. of Twente, 7500 AE Enschede, The Netherlands)

We present a new optical characterization of the behavior of single ultrasound contrast bubbles. The method consists of insonifying individual bubbles several times successively sweeping the applied frequency and recording movies of the bubble response up to 25 million frames per second with an ultra-high-speed camera operated in a segmented mode. The method, termed microbubble spectroscopy, enables one to reconstruct a resonance curve of the oscillating bubble in a single run. We analyze the data through a linearized model for coated bubbles. The results confirm the significant influence of the shell on the bubble dynamics: shell elasticity increases the resonance frequency by about 50%, and shell viscosity is responsible for about 70% of the total damping. The obtained value for shell elasticity is in quantitative agreement with previously reported values. The shell viscosity decreases significantly with the dilatation rate, revealing the nonlinear behavior of the phospholipid coating.

9:20

5aBB2. High-frequency response of polymer shell contrast agents. Jeffrey A. Ketterling, Jonathan Mamou (Lizzi Ctr. for Biomed. Eng., Riverside Res. Inst., 156 William St., New York, NY 10038), John S. AllenIII (Univ. of Hawaii-Manoa, Honolulu, HI 96822), Orlando Aristizábal, Rene G. Williamson, and Daniel H. Turnbull (NYU School of Med., New York, NY 10016)

Ultrasound contrast agents were originally designed for use at low MHz frequencies; however, recent experimental evidence suggests that many of these agents may also be useful for contrast imaging at frequencies above 20 MHz. Three polymer shelled agents (mean diameters of 0.56, 1.1, and 3.4 μm) from POINT Biomedical were investigated to determine their optimal high-frequency responses. A flow phantom was constructed to restrict the flow of a dilute contrast agent solution to a small volume. The focus of a 40-MHz single-element transducer from a Visual Sonics V770 was positioned within the flow phantom. Radio-frequency backscatter data from individual contrast agents were digitized under a variety of pulse durations (1–15 cycles) and pressure levels (1.4–7.6 MPa). Echo signals from individual bubbles were windowed, and spectra were calculated. Each of the POINT agents reflected energy at the 40-MHz fundamental; however, only the 1.1-μm-diameter agent displayed characteristics of a harmonic response under longer excitations (15 cycle) and medium pressure settings (2.5 to 5 MPa). Experimental results were com-

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pared to theoretical calculations for the case of the polymer shell remaining intact during excitation and for free gas nuclei that result from a ruptured shell.

9:35

5aBB3. Optical observations of the collapsing behavior of polymer micro-capsule irradiated in the ultrasonic sound field. Tetsuo Yamato, Kenji Yoshida, Wataru Kiyan, and Yoshiaki Watanabe (Faculty of Eng., Doshisha Univ., 610-0321 1-3 Tataramiyakotani Kyotanabe Kyoto Japan)

For the medical use, it is important to understand the micro-capsule behavior during collapsing under the ultrasonic sound field. In this report, the behavior of micro-capsule collapse was observed using the high-speed video camera system (HPV-1, SHIMADZU: maximum recording rate; 1 000 000 frames/second). By using this system, the detail phenomena can be observed. The micro-capsule leaked the internal gases through a small pin hole on the surface of the micro-capsule while repeating asymmetrical vibration. The internal gases were emitted during the positive period of driving sound pressure and returned again in the micro-capsule during the negative period. By repeating the outflow and inflow of internal gases, the stress was cumulated on the shell. Therefore, the small pin hole developed into the big one, then, the micro-capsule emitted a large amount of internal gases and the capsule was collapsed. After the collapsing, only the shell was remained. From the results, it was found for the effect of the shell, that is, the micro-capsule radius shrunk in positive pressure phase, however it hardly expanded in negative pressure phase. These phenomena were numerically simulated by the vibration model, which the effect of the shell was removed only when shrinking. This simulation results qualitatively agreed with observation results.

9:50

5aBB4. Thermal effect of microbubbles in the focused ultrasound field. Yukio Kaneko, Naoyuki Iida, Shu Takagi, and Yoichiro Matsumoto (Dept. of Mech. Eng., The Univ. of Tokyo, Tokyo, Japan)

High-intensity focused ultrasound (HIFU) has been developed for the treatment of tumor, and additionally the medical applications with microbubbles such as ultrasound imaging have attracted much attention. In the field of the bubble dynamics, it is known that the microbubble is an energy converter of ultrasound mechanical energy to heat when subjected to an acoustic field. The goal is the tissue heating enhanced by microbubbles. In this study, the relationship between the heating effect and the behavior of microbubbles is analyzed. The temperature rise was measured by a thermocouple and a thermal liquid crystal sheet, and the bubble behavior was simultaneously observed with a camera. The ultrasound frequency is 2.2 MHz and the intensity is 100–2000 W/cm^2. The temperature rise became larger as the number density of microbubbles (Levovist) around the focal region increased. As the number density became too large, the conversion efficiency from ultrasound energy to heat became smaller because the shielding effect of ultrasound by the existence of too many bubbles became larger. From these results, it is indicated that it is important to choose the optimal conditions of microbubbles and ultrasound in order to control the heat deposition from microbubbles.

10:05–10:25 Break

10:25

5aBB5. Engineering the acoustic response of lipid-based microbubbles: Simulations and high speed optical imaging results. Eleanor Stride (Dept. of Mech. Eng., UCL, Torrington Pl., London WC1E 7JE, UK, e_stride@meng.ucl.ac.uk), Rob Eckersley, Kevin Chetty, Charles Sennoga, and Jo Hajnal (Imperial College, London, W12 0NN, UK)

Gas microbubbles coated with a surfactant or polymer shell have become well established as contrast agents for ultrasound imaging and are under investigation for therapeutic applications such as targeted drug delivery. The purpose of this study was to examine the effect of modifying the microbubbles' coating composition upon their response to ultrasound excitation. Using a high speed camera operating at 3 MHz, the radial oscillations from the commercial contrast agent SonoVueTM and four in-house microbubble preparations were measured under controlled insonation conditions (four-cycle Gaussian pulse; center frequency 0.5 MHz; peak negative pressure 30–70 kPa). The results indicate that microbubble response varies as the lipid structure and functional components in the coating change. These findings have been compared with simulation results using the generalized model for microbubble dynamics developed by the authors, in order to quantify the observed effects of changing the shell composition upon the microbubbles viscoelastic and diffusion characteristics. Both microbubble shrinkage and variable shell viscoelasticity can have significant implications for the acoustic response. This in turn can affect contrast specific imaging methods, especially those involving multipulse sequences such as pulse inversion. Similarly, these phenomena may affect the microbubble destruction threshold and hence the efficiency of drug delivery procedures.

10:40

5aBB6. The use of homemade microbubbles in detection of organ bleeding and enhancement of high-intensity focused ultrasound therapy. Wenbo Luo, Vesna Zderic, and Shahram Vaezy (Dept. of Bioengineering, Univ. of Washington, Seattle, WA 98195)

We investigated the feasibility of using homemade microbubbles in detection of internal bleeding, and for optimization of high-intensity focused ultrasound (HIFU) therapy. Homemade microbubbles were produced by sonicating albumin and dextrose saline solution with the presence of perfluorocarbon gas. Microbubbles were injected intravenously to reveal rabbit kidney and liver injuries. HIFU treatments were performed on liver lacerations for hemostatis and on intact liver for lesion production, with or without homemade microbubbles. The mean size of homemade microbubble was 4.7 um (range 1.2–7.1 μm). The concentration of microbubbles was 3.1*108 bubble/ml. Kidney and liver injuries were revealed by mosaic color flow at the injury sites. In treatment, the average hemostasis times normalized with the bleeding rate were 39 s/ml*min for HIFU only and 21 s/ml*min for HIFU+microbubbles treatments. The presence of microbubbles reduced 44 percent ($p<0.05$) time in producing the first formation of the blood coagulum. The HIFU lesion depth was reduced by 35 percent because microbubbles shield the postfocal region from HIFU exposure. Homemade microbubbles oscillate and collapse in ultrasound field, generating mosaic pattern thus revealing the bleeding site. It also shows potential in promoting HIFU hemostasis and shielding deep-tissue regions from HIFU exposures.

10:55

5aBB7. The effects of encapsulation on growth and dissolution of a contrast microbubble. Kausik Sarkar and Pankaj Jain (Mech. Eng., Univ. of Delaware, Newark, Delaware, sarkar@me.udel.edu)

Micron-size gas bubbles are intravenously injected into patients body at the time of ultrasound imaging to improve image contrast. The bubbles are encapsulated by a thin layer (4–10 nm) of protein, lipids, and other surface active materials, to prevent their premature dissolution in the blood. We will present a model for the dissolution of the microbubble that accounts for the effects of encapsulation. The encapsulation hinders the permeability of the gas-liquid surface and its elasticity balances the surface tension-induced stress. Both these effects will be explicitly modeled. The model behavior will be discussed for variations of the material parameters and conditions (encapsulation permeability and elasticity, mole fraction of the osmotic agent and liquid saturation). The encapsulation significantly affects the bubble growth and dissolution including their time scales.

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Session 5aEA

Engineering Acoustics: Development and Application of Practical Electroacoustical Devices

Juan I. Arvelo, Cochair
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Contributed Papers

8:00

5aEA1. Wideband piezoelectric rectangular loudspeaker using a tuck-shaped polyvinylide-fluoride bimorph. Taira Itoh, Juro Ohga (Shibaura Inst. of Technol., 3-7-5, Toyosu, Koto-ku, Tokyo 135-8548, Japan, m106011@shibaura-it.ac.jp), Toshitaka Takei (Take T Co.), and Nobuhiro Moriyama (Kureha Co.)

A bimorph sheet of polyvinylide-fluoride (PVDF) film was applied to a flat rectangular loudspeaker as a folded zigzag-tack shape diaphragm whose size is, e.g., 260–144 mm with various depths. These loudspeakers, which we call bimorph tuck loudspeakers, are characterized by their moderate size, wide frequency range, light weight, and lack of magnetic flux radiation. This study examines the electro-acoustic transducer characteristics of these loudspeakers. Their sensitivity and resonance frequency were measured using a flat-panel baffle. Their electrical properties and sound field characteristics were also examined. A speaker's resonance frequency was estimated theoretically as a function of tuck depth. The measured resonance frequencies were, for example, 200, 160, 90, and 60 Hz. These low resonance frequencies are satisfactory for full-range loudspeaker use. Frequency characteristics of output sound pressure were also measured. They still include some irregularities that are attributable to a local bending resonance of the diaphragm. This paper examines the control methods for those irregularities.

8:15

5aEA2. Digital measurement method for dynamic distortion of loudspeakers. Keiichi Imaoka and Juro Ohga (Shibaura Inst. of Technol., 3-7-5 Toyosu, Koutou-ku, Tokyo, 135-8548, Japan, m605101@sic.shibaura-it.ac.jp)

Though many sorts of measuring methods using digital signal processing techniques have been developed, most of them are only for measurements in a linear range. There is still no suitable digital measuring method for nonlinear distortion of acoustical devices. An accurate and convenient nonlinear distortion measurement system should be developed for evaluation of electroacoustical transducers. A new digital distortion measuring method for acoustical devices is presented briefly. This method applies a Pink-TSP signal (time stretched pulse, i.e., quickly swept sinusoidal signal), whose frequency band is partially eliminated, to an acoustical system to be measured. The detected component produced in the rejected band is picked up by a bandpass filter and measured as a distortion. First, two kinds of small size electrodynamic loudspeaker units were measured by using the dynamic distortion measurement system mentioned above. Next, distortion characteristics of mechanical single resonant system models whose component shows amplitude-dependent nature were examined. This paper describes the relationship between distortion characteristics and quality factor, input signal level, and nonlinear parameters. The difference in distortion of two kinds of loudspeakers is detected by this measurement. It is seen that the resonant phenomenon has an influence on distortion characteristics.

8:30

5aEA3. Effect of acoustical load parameters for frequency response of headphones. Megumi Kobayashi, Juro Ohga, Wataru Onoda (Shibaura Inst. of Technol., 3-7-5 Toyosu, Koto-ku, Tokyo, 135-8548 Japan), and Ikuo Oohira (Ashida Sound Co., Ltd. Shinagawa-ku, Tokyo, 141-0032 Japan)

This report describes experimental and theoretical analyses for characteristics of headphones to examine the effect of significant parameters for design. Though various types of headphones are being used, the standard design procedure is still not established, because physical relationship between headphone construction and the human ear is not understood satisfactorily. The authors measured frequency response of various headphones by using IEC 60318 artificial ear and IEC 60959 head and torso simulator (HATS), and also measured the response by human ears by inserting a miniature microphone for comparison. By these examinations, a few mechanoacoustical parameters important for design were picked out. Especially, the effect of acoustical leakage shall be noted. Design parameters for supra-aural and intraconcha headphones will be reported.

8:45

5aEA4. High-output microsized earplug driver system for extreme acoustic environments. Keehoon Kim, Andrew Kostrzewski (Physical Optics Corp., 20600 Gramercy Pl., #100, Torrance, CA 90501, kkim@poc.com), and Reginald Daniels (Air Force Acoust. Res. Lab., Wright–Patterson AFB, OH 45433)

Hearing protection is critical when aircraft flight operations create a high-noise environment (130–150 dB) for aircrew and service members. Untreated aircraft noise causes noise-induced hearing loss (NIHL) and poses risks of similar hearing damage in the future. The authors have developed a new microsized smart material actuator (MSMA), a high-output earplug driver for active noise reduction (ANR) hearing protection, based on highly efficient microstructured acoustic exciters. The MSMA consists of a microscale earplug (MEP) with a proprietary acoustic structure, a protective MEP package, and a compact actuation amplifier easily connected to an external ANR controller. Performance tests show that the system produces over 130 dB into a 1-cc trapped cylindrical volume, 6 mm in diameter by 7 mm long, without significant phase delay or noticeable discontinuities in frequency response.

9:00

5aEA5. The surface and bulk microfabrication of optical seismometers and vibrometers using Sandia National Laboratories' silicon micromachining technology. Murat Okandan, Neal A. Hall, Robert Littrell (Sandia Natl. Labs., Albuquerque, NM 87185), Baris Bicen, and F. L. Degertekin (Georgia Inst. of Technol., Atlanta, GA 30332)

The microfabrication of optical seismometer and vibrometer structures using Sandia's silicon-based process technology is presented. The structure employs a 1.5-mm-diameter polysilicon diaphragm anchored to a

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200- to 400-μm-thick bulk silicon proof mass, and suspended by surface micromachined polysilicon springs. Motion of the diaphragm is detected using an integrated polysilicon surface micromachined grating-based optical interferometer. Initial dynamic characterization of these structures in vacuum shows a fundamental resonant frequency of 250 Hz. Based on the high force sensitivity of these structures combined with interferometric motion detection, 1-μg/$\sqrt{\text{Hz}}$ detection resolution in the 1-Hz frequency range is estimated. Microfabricated structures employing electrostatic actuation electrodes for the diaphragm are also presented, which enable force feedback operation for high dynamic range and enhanced dc stability. These characteristics may make these sensors well-suited for seismic and inertial navigation applications.

9:15

5aEA6. Finite-element modeling of thin-film damping in micromachined microphones employing diffraction based optical readout. Neal A. Hall, Murat Okandan (Sandia Natl. Labs., Albuquerque, NM 87185), and F.L. Degertekin (Georgia Inst. of Technol., Atlanta, GA 30332)

Micromachined microphones with diffraction-based optical displacement detection have been presented in detail previously [J. Acoust. Soc. Am. **118**(5), 3000–3009 (2005)]. In addition to providing superior diaphragm displacement detection sensitivity over miniature capacitive sensors, optical approaches have the advantage of removing mechanical design constraints on the microphone backplate perforation architecture as well as constraints on the diaphragm-backplate gap thickness. Taking advantage of this freedom to tailor design the frequency response function and thermal noise characteristics of miniature high-performance microphones requires a sophisticated damping model to navigate this new design space. A finite-element model in ANSYS based on the modal projection method is employed to study the dynamics of new optical microphone structures. The model extracts the frequency-dependent resistance and stiffening characteristics of the film using modal displacement profiles of the diaphragm. Simulated frequency response functions and thermal-mechanical noise limits agree well with those measured on fabricated structures. Most notably, 1.5-mm-diameter diaphragm structures with under 1 μPa/$\sqrt{\text{Hz}}$ thermal noise and over 20-kHz bandwidth have been successfully designed, fabricated, and characterized. [The authors would like acknowledge the IC Postdoctoral Research Fellowship Program.]

9:30

5aEA7. Design optimization of a microelectromechanical piezoresisitive microphone for use in aeroacoustic measurements. Brian Homeijer, Ben Griffin, Toshi Nishida, Lou Cattafesta, and Mark Sheplak (Univ. of Florida, 231 MAE-A Bldg., Gainesville, FL 32611-6250, sheplak@ufl.edu)

A microelectromechanical systems (MEMS)-based piezoresistive microphone optimimum design is presented, with a focus on improving the minimum detectable pressure over many current technologies without sacrificing bandwidth. This microphone design addresses many of the problems associated with previous piezoresistive microphones. Here, a novel nonlinear circular composite plate mechanic model was employed to determine the stresses in the diaphragm, which was designed to be in the compressive quasibuckled state. With this model, the inherent in-plane stresses that occur in the microelectronic fabrication process can be used to increase the sensitivity of the device. Ion-implanted doped silicon was chosen for the piezoresistors and a fabrication recipe was made which minimizes the inherent noise characteristics of the material. The piezoresistors are arranged in a Wheatstone bridge configuration with two resistors oriented for tangential current flow and two for radial current flow. A lumped element model was created to describe the dynamic characteristics of the microphone diaphragm and the cavity/vent structure. The geometry for this device was optimized using a sequential quadratic programming scheme performed using the aforementioned novel device characteristics. Results indicate a dynamic range in excess of 120 dB for devices possessing resonant frequencies beyond 120 kHz.

9:45

5aEA8. A two-dimensional mechanical resonator array filter with reduced sensitivity to disorder. John A. Judge, Joseph F. Vignola (Mech. Eng. Dept., The Catholic Univ. of America, Washington, DC 20064, judge@cua.edu), and Scott A. Mathews (The Catholic Univ. of America, Washington, DC 20064)

The structural acoustics of perfectly periodic arrays of coupled mechanical resonators are characterized by frequency passbands, outside of which waves do not propagate across the array, but in which wave propagation is attenuated only by dissipation. Arrays of high-Q micromechanical resonators, with sufficiently high resonant frequencies, can thus be used as filters for rf band electrical signals. However, small deviations from periodicity, such as the disorder induced by finite fabrication tolerances, are known to have significant consequences for energy propagation across such arrays. This is particularly true in narrow passband applications, in which resonators are weakly coupled, and the passband energy transmission can be significantly degraded. In this work, a two-dimensional array of resonators is demonstrated that is less sensitive to disorder than equivalent one-dimensional arrays, improving filter performance without the need for improved manufacturing tolerances. The concept is explained in terms of a simple model, and the design of a prototype array is described, including a statistical investigation of disorder effects using finite-element analysis. Experimental measurements are reported demonstrating improved performance of the two-dimensional array (relative to one-dimensional arrays), by comparing laser Doppler vibrometry measurements of equivalent 2-D and 1-D arrays created in silicon wafers using laser microfabrication.

10:00–10:15 Break

10:15

5aEA9. Methods of fitting a nonaudible murmur microphone for daily use and development of urethane elastmer duplex structure type nonaudible murmur microphone. Yoshitaka Nakajima and Kiyohiro Shikano (Grad. School of Information Sci., Nara Inst. of Sci. and Technol., 8916-5 Takayama, Ikoma, Nara, 630-0101 Japan)

Nonaudible murmur (NAM) microphones capturing vibrations directly from the skin without mediation by the air enable the sampling of nonaudible murmur with almost the same amplification rate as normal speech. Because they are filtered through the human body, they are robust against air-conducted noises. Soft silicone has been used, which has acoustic impedance close to human flesh, as a sound medium between the electrode of the condenser microphone and the skin by fixing it in a neckband style. We replaced the soft silicone with urethane elastomer, which has self-adhesiveness, to develop a NAM microphone that is easily wearable in everyday life without expendable supplies or supporting devices. It can be fixed to the skin, and it also decreases body movement noises. But miniaturized or sticking-type NAM microphones increased the mixture of air-conducted noises from the back of the microphone and from a gap between the skin and the rim of the microphone. To solve this problem, we created a duplex structure inside the urethane elastomer wrapping condenser microphone and outside the urethane elastomer for fixation and noise proofing to develop a miniaturized sticking-type NAM microphone that was again robust against air-conducted noise.

10:30

5aEA10. Throat-contact microphone with an elastomer-supported piezoelectric diaphragm. Daisuke Yoshioka, Hiroaki Okano, Juro Ohga (Shibaura Inst. of Technol., 3-7-5, Toyosu, Koutou-ku, Tokyo, 135-8548, Japan), Ikuo Oohira (Ashida Sound Co., Ltd., Shinagawa-ku, Tokyo, 141-0032, Japan), and Masato Miyoshi (NTT Commun. Sci. Labs., Seika-cho, "Keihanna Sci. City," Kyoto, 619-0237, Japan)

Throat-contact microphones are free from environmental acoustical noise because they pick up only local vibration of the throat. This paper proposes a new piezoelectric throat-contact microphone using an elastomer-supported diaphragm. It is smaller and lighter than conventional

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electromagnetic throat-contact microphones, which have a magnet and coil. Transducer characteristic and parameters are examined in this paper to establish design procedures for such microphones. Frequency characteristics of output signals and electrical impedance for the microphones and their transducing components are presented. The piezoelectric diaphragm is supported by a few elastomer boundaries.

10:45

5aEA11. A study of bone conduction and its applications for audio signal transmission. Yusuke Watabe, Kochi Yui, Yuko Watanabe, and Hareo Hamada (Tokyo Denki Univ., 2-1200 Muzai-gakuendai, Inzai, Chiba Prefecture 270-1382, Japan)

Human can perceive a sound in two different ways: one is sound transmission through the ear canal, another is that through the skull. The bone conduction technology has been used mainly for hearing impaired listeners and as communication tools in the military organization. Recently, because of a rapid improvement of the bone conduction device, its technology has spread in the field of audio. However, the relationship between the physical characteristic and the subjective responses of bone conduction are not clearly understood. For example, the effect of attachment position and excitation force of bone conduction loudspeakers has been still under discussion. In addition, since the bone conduction units also generate the airborne sound, listeners can hear the sound composed of both the bone conduction sound and the airborne sound. In our previous work, by using of a loudness curve and a threshold of hearing, a fundamental performance of the bone conduction was discussed. In this paper, further investigation will be carried out in order to establish a transmission system of the bone conduction, and the airborne sound on the eardrum that is caused by the vibration of the bone conduction loudspeaker will be measured.

11:00

5aEA12. Acoustic communication system using bone conduction elements. Kenji Muto (Tokyo Metropolitan College of Industrial Technol., 8-52-1 Minamisenju, Arakawa-ku, Tokyo, Japan, muto@kouku-k.ac.jp), Guoyue Chen (Akita Prefectural Univ.), Kunihiko Takano (Tokyo Metropolitan College of Industrial Technol.), Kikuo Asai, and Kimio Kondo (Natl. Inst. of Multimedia Education)

This study examines an acoustic communication system constructed using a bone conduction speaker and a bone conduction microphone. This system is a headset that places a bone conduction speaker to the user's cheek, and a bone conduction microphone to the other cheek. A user can thereby hear a voice through the bone conduction speaker while hearing a voice of a loudspeaker and surrounding noise when using it for teleconferencing. It can transmit the voice of the remote loudspeaker using the external ear, but the user can hear the translated voice through bone conduction. However, the sound of the bone conduction speaker and the surrounding sound are mixed with the speaker's voice using the bone conduction microphone. The vibration level of the bone conduction microphone from the bone conduction speaker was about 0 dB, although it was the same level as the level of the speaker's voice. The vibration level from surrounding sound was about −3 dB, although it was a smaller level than the level of the speaker's voice. This device removed mixed sounds using the echo canceller [K. Muto *et al.*, Proc. Inter-noise (2004), p. 248]. The echo elimination level was about 12 dB. [This work was supported by JSPS KAKENHI 17300283.]

SATURDAY MORNING, 2 DECEMBER 2006

IAO NEEDLE/AKAKA FALLS ROOM, 8:20 TO 11:35 A.M.

Session 5aMU

Musical Acoustics: Simulation and Measurement Techniques for Musical Acoustics I

Murray D. Campbell, Cochair
Univ. of Edinburgh, School of Physics, Mayfield Rd. Edinburgh, EH9 3JZ, U.K.

Gary P. Scavone, Cochair
McGill Univ., Music Technology, Schulich School of Music, 555 Sherbrooke St., W., Montreal, Quebec, H3A 1E3, Canada

Shigeru Yoshikawa, Cochair
Kyushu Univ., Dept. of Acoustic Design, 4-9-1 Shiobaru, Minami-Ku, Fukuoka 815-8540, Japan

Invited Papers

8:20

5aMU1. Observing the effects of waveguide model elements in acoustic tube measurements. Tamara Smyth (Simon Fraser Univ., Surrey, BC, Canada) and Jonathan Abel (Universal Audio Inc., Santa Cruz, CA)

The theory of digital waveguide synthesis and its use in modeling virtual musical instruments, and in particular for cylindrical and conical bores, is well documented. Current models rely on certain approximations, however, particularly in cases where the theory provides no exact closed form solution, such as the reflection and transmission occurring at the bore's open end. In this research, we observe, from a time domain, waveguide model perspective, how the theory corresponds to actual acoustic measurements. We consider four simple acoustic tube structures, incorporating both open and closed boundary conditions for both a simple cylinder and a cylinder with a conical flare, allowing us to isolate, and observe, the filtering effects of each model component.

5a SAT. AM

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8:40

5aMU2. Ingoing and outgoing waves in time-domain simulation of wind instruments. Eric Ducasse (Laboratoire de Mecanique Physique, UMR CNRS 5469, Universite Bordeaux I, 351 cours de la Liberation, F-33405 Talence Cedex, France)

An extension of a recently published work [Ducasse, J. Acoust. Soc. Am. **112**, 3031–3041 (2002)] is presented. Two-port modeling is an efficient tool for time-domain simulation of sound propagation in wind instruments. It requires the separation of ingoing and outgoing waves contributing to the acoustic field in any cross section of the air column. This separation is achieved by using a multimodal approach, where uncoupled traveling modes appear in the case of cylindrical waveguides: the propagation of the plane mode is straightforwardly modeled by a delay line and the impulsional response of any other mode is the sum of a delayed Dirac pulse and a closed-form dispersive term. Although mode coupling occurs in acoustic pipes with nonuniform cross section, the scattering matrix can be directly obtained. When a multimodal approach is not adequate, the time-domain responses of complex elements are numerically calculated by introducing either nonreflecting boundary conditions (NRBCs) or perfectly matched layers (PMLs) in a finite-element method. The examples of (*i*) the reflection matrix of a bell, as seen from the body, and (*ii*) a piece of pipe containing an obstacle (e.g., the register hole) illustrate the method.

9:00

5aMU3. Differences between cylindrical and conical brass instruments; the nonlinear propagation point of view from experiments and simulations. Joel Gilbert (Laboratoire d'Acoustique de l'Universite du Maine, CNRS, avenue Olivier Messiaen, 72085 Le Mans Cedex 9, France, joel.gilbert@univ-lemans.fr)

The brightness of the sound generated by brass instruments is due to the essential nonlinearity of the wave propagation in the pipe. The bright instruments such as the trumpet and the trombone are different from brass instruments such as saxhorns or flugelhorns. The bright instruments have a cylindrical pipe segment just downstream of the mouthpiece. The conical bore of the saxhorns implies a faster decay of the wave, which reduces the nonlinear wave steepening. In this talk, theoretical and experimental results of brassiness associated with brass instruments having different bores, from mainly cylindrical ones to mainly conical ones, will be shown. The theoretical results come from simulations of periodic regimes using a theoretical background based on generalized Burger's equations.

9:20

5aMU4. Motion of brass players' lips during very loud playing. Seona Bromage, Murray Campbell, Samuel Stevenson (School of Phys., Univ. of Edinburgh, Edinburgh EH9 3JZ, UK, d.m.campbell@ed.ac.uk), John Chick (Univ. of Edinburgh, Edinburgh EH9 3JL, UK), and Joël Gilbert (Laboratoire d'Acoustique de l'Université du Maine, 72085 Le Mans Cedex9, France)

When a crescendo is played on a trumpet, horn, or trombone, the timbre changes markedly at the fortissimo level: there is a dramatic increase in the relative amplitudes of the high- frequency harmonics, and the resulting sound is often described as ''brassy.'' Shock-wave generation in the air column of the instrument as a consequence of nonlinear steepening of the wavefront is the generally accepted explanation of this phenomenon. It has, however, also been suggested that saturation of the lip opening could play a role in the timbre change. This study compares the motion of the lips at lower dynamic levels and during extremely loud playing, using a high-speed digital camera and instruments with different sizes of mouthpiece. Curves illustrating the variation of the open lip area as a function of time at different dynamic levels are presented, together with waveforms and spectral analyses of both the radiated sound and the pressure in the mouthpiece. The results suggest that there is no qualitative change in the time-dependent behaviour of the opening area of the lips when the instrument enters the brassy regime.

9:40

5aMU5. Input impedance measurements of alto saxophones with a calibration error analysis. Antoine Lefebvre and Gary Scavone (Music Tech. Area, McGill Univ., Montral H3A 1E3, Canada)

Recent investigations are presented for the measurement of acoustic input impedance of wind music instruments. Initial efforts have focused on measurements of alto saxophones using a two-microphone, three-calibration tube procedure. We first present the design of the impedance probe, as well as the data sampling and analysis procedures. We then discuss the relationship between errors in the calculated calibration parameters and the resulting measured transfer functions. Finally, we compare the input impedances of Yamaha Z and Selmer Series II alto saxophones for several low register fingerings. We note several instances where the Yamaha saxophone possesses significantly stronger fundamental resonances than the Selmer saxophone.

10:00–10:20 Break

Contributed Papers

10:20

5aMU6. Interactive program for computer-aided design of woodwind musical instruments. Héctor Alfonso Cordourier-Maruri and Felipe Orduña-Bustamante (CCADET-UNAM, Circuito Exterior CU, CP 04510, México DF, México, felipe@aleph.cinstrum.unam.mx)

Current scientific knowledge on the acoustics of woodwind instruments allows accurate physical modeling of the sound generation mechanisms, and the acoustic interaction with the air column. These physical models can be incorporated into computer software, in order to evaluate not only the tuning, voicing, and other tonal characteristics of existing instrument designs, but also to estimate the potential effects of proposed modifications, and to anticipate the characteristics of new designs. This presentation describes the development of an interactive computer program for the acoustic design, evaluation, and auralization of woodwind musical instruments. The program displays a graphical user interface that allows convenient entry of air column data, including constant and variable cross-section segments, open and closed fingerholes, and air column terminations. It also implements different types of sound excitation mechanisms, which include jet-driven (flutelike) and single reed (clarinetlike) models. The program can produce input impedance and reflection function data for further acoustic analysis, and synthetic sounds that approximate the musical tones that can be expected form the actual instru-

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ment. The software has been validated by comparative analysis of calculated versus measured air columns, and simulated versus actual (recorded) flute tones.

10:35

5aMU7. Control parameters inversion using genetic algorithms applied to numerical impedance synthesis for woodwinds. Laura Perichon, Olivier Carriere, Jean-Pierre Hermand, Matthias Meyer (Environ. Hydroacoustics Lab, Optics and Acoust. Dept., ULB-CP 194/05, 50 Av. F.D. Roosevelt, B-1050 Brussels, Belgium), and Philippe Guillemain (Ctr. Natl. pour la Recherche Scientifique, 13402 Marseille cedex 20, France)

Woodwind sound synthesis in real time has been achieved using different computational techniques, especially the so-called numerical impedance model based only on acoustical variables. The impedances of the excitation mechanism and of the resonator are approximated by IIR filters and related by a nonlinear coupling. The control parameters can be reduced to a number of three including the length of the resonator, the mouth pressure, and a mask parameter. In this paper we present preliminary inversion results for the latter two parameters in the case of a clarinet. The inversion algorithm is based on differential evolution metaheuristics with a number of prior conditions on the correlated parameters. A cost function based on time-domain correlation has been implemented, allowing us to obtain better results in terms of signal matching than with conventional descriptors extraction. The technique has proved its effectiveness on the sustained part of synthesized sounds and a regularization method is presented to allow us to invert attack and decay parts as well as real analogically recorded signals. Results will be of main interest for tuning control parameters in real-time model-based synthesis.

10:50

5aMU8. Virtual wind instruments based on pulse forming synthesis. Michael Oehler (Inst. for Appl. Musicology and Psych., Duerener Strasse 98, 50931 Cologne, Germany, oehler@iamp.info) and Christoph Reuter (Univ. of Cologne, 50923 Cologne, Germany)

A new synthesis method for wind instruments is presented, the digital pulse forming. The core of that principle is that every wind instrument sound can basically be put down to its excitation impulses, which independently of the fundamental always behave according to the same principles. By controlling the pulse width and shape it is possible to generate all sound nuances that can be produced on a real wind instrument. Based on the 1975 found principles of generating wind-instrument-like spectra with typical stable formant areas and spectral gaps evoked by the excitation pulses of double-reeds or lips [J. P. Fricke, Fortsch. Akust. **4**(DAGA75), 407–411 (1975)], a virtual wind instrument, the Digital Variophon, is developed. The resulting software-based version of the original Variophon, the first analog wind synthesizer using the pulse forming principles, is a further step towards the intended scientific experiment system for analyzing and synthesizing (wind) instrument sounds. Trumpet, bassoon, and oboe modules are already digitally implemented in a flexible, realtime sound generating and analyzing framework. Acoustic analyses and hearing experiments show extensive concordance between the original instruments and the Digital Variophon. [Work supported by DFG.]

11:05

5aMU9. Flow regimes at the output of oboe double-reeds. Christophe Vergez (LMA, CNRS UPR-7051, 31 chemin Joseph-Aiguier, 13402 Marseille, cedex 20, France) and Reneé Caussé (IRCAM)

Measurements of the quasistatic nonlinear characteristic curve for double reeds show systematic deviations from an elementary model of a reed instrument based on a spring model associated to a Bernoulli flow input into the reed. These deviations were previously explained by the pressure recovered by the flow as it expands in the conically divergent output duct of the reed. We will show measurements of the pressure recovery inside the reed as well as detailed flow profiles that can explain the different regimes observed for the pressure recovery (laminar and turbulent). An extension of these measurements to oscillating regimes allows us to observe some qualitative differences in the flow, suggesting an important influence of the dynamic terms in the Navier-Stokes equation in these kinds of regimes.

11:20

5aMU10. Wolf notes in brass wind instruments. Thomas R. Moore and Isaac E. W. Codrey (Dept. of Phys., Rollins College, Winter Park, FL 32789)

It has been recently reported that experiments using artificial lips have shown that damping the bell vibrations of a trumpet can result in changes in the sound spectrum. New data now suggest that these effects are enhanced when the fundamental frequency of the note is near a resonance frequency of the bell vibration. This effect results in a situation that is analogous to the wolf note commonly found on stringed instruments. Although the physics of the process is still poorly understood, the effect appears not to be related to a change in impedance of the air column.

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Session 5aNS

Noise and Physical Acoustics: Prediction and Propagation of Outdoor Noise I

D. Keith Wilson, Cochair
U.S. Army Cold Regions Research Lab., Engineering Research and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290

Kohei Yamamoto, Cochair
Kobayashi Inst. of Physical Research, 3-20-41 Higashi-Motomachi, Kokubunnji-shi, Tokyo 185-0022, Japan

Chair's Introduction—7:30

Invited Papers

7:35

5aNS1. Sound field modeling in a street canyon with a diffusion equation. Judicaël Picaut, Stéphane Colle, and Michel Bérengier (LCPC, Section Acoustique Routière et Urbaine, Rte. de Bouaye, BP 4129, 44341 Bouguenais Cedex, France)

The transport theory of sound particles is applied to the sound field modeling in an empty street canyon with partially diffusely reflecting facades. A diffusion equation is then derived to predict the sound field distribution and the sound decay in the street. The main parameter, namely the diffusion coefficient, is a function of the street width and the acoustic reflection laws of both building facades. For a single rectangular street, analytical solutions can be found. For more complex urban spaces, a finite-element based approach is proposed. Several numerical examples are given, like the sound propagation in a street with a nonuniform cros section, and in street intersections, for point and line sources. In comparison with ray-tracing-based models, the diffusion model requires less computation time and could be applied to the calculation of a sound map for large urban areas. However, at the present time, the diffusion coefficient can be calculated only for simple building reflection laws, like the Lambert's one.

7:55

5aNS2. Prediction of road traffic noise using two-dimensional numerical analysis. Shinichi Sakamoto (Inst. of Industrial Sci., The Univ. of Tokyo, 4-6-1 Komaba, Meguro-ku, Tokyo 153-8505, sakamo@iis.u-tokyo.ac.jp), Akinori Fukushima (NEWS Environ. Design, Inc., Hyougo, Kobe 652-0802), Tomonao Okubo, and Kohei Yamamoto (Kobayasi Inst. of Physical Res., Kokubunji, Tokyo 185-0022, Japan)

For acoustically complicated road structures such as semi-underground roads and special areas in which a viaduct road and a flat road with noise barriers exist together, prediction of road traffic noise is complicated because of multiple reflections and diffractions that occur inside the road structures. For such road structures, an energy-based engineering model cannot be applied and noise propagation should be addressed through introduction of wave theory. When the road structures have almost identical cross-sectional shape along the road, two-dimensional (2-D) wave-based numerical analyses are applicable. In the prediction model of road traffic noise, the ASJ-RTN Model 2003, published by the Acoustical Society of Japan (ASJ), the application of 2D wave-based numerical analysis was introduced as a prediction method for such complicated road structures. Comparisons between calculations by BEM and FDM and field measurements and experiments for three actual road structures were conducted. Consequently, calculation results agreed well with measured ones. Therefore, the validity of the calculation methods was confirmed. This research was discussed in the Research Committee of Road Traffic Noise in ASJ. Measurement data were provided by Nippon Expressway Company and the Nagoya Expressway Public Corporation.

8:15

5aNS3. Modeling outdoor sound propagation in mountainous areas. Dick Botteldooren, Timothy Van Renterghem, and Bram de Greve (Acoust. Group, Dept. of Information Technol., Ghent Univ., St. Pietersnieuwstraat 41, 9000 Gent, Belgium, dick.botteldooren@ugent.be)

Detailed modeling of outdoor sound propagation in mountainous area imposes special requirements. The strong gradients in ground surface create propagation conditions that may lead to noise levels much higher than expected at some distance from the source partly due to a focusing effect. Meteorological conditions are particular: main winds following the valley, strong temperature gradients often including stable inversion, slope winds, etc. Because of these particular conditions, efficient modeling of outdoor sound propagation in mountainous area requires adaptations to be made to general purpose models. In this paper we will present possibilities for making time-domain models terrain following and consider options to decrease the memory requirements. For the higher frequency range and distances one is generally interested in, these adaptations are insufficient. Hence, a Green's function parabolic equation model is added to the toolbox. To tackle the problem in three dimensions, a number of two-dimensional slices connecting the receiver point to all (segments of) sources considered is made. Propagation results for each slice are added. Numerical simulations are compared to field measurements made close to an Alpine highway.

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8:35

5aNS4. Measurements of acoustic impedance and their application to sound propagation calculations and audible simulations. Teruo Iwase (Faculty of Eng., Niigata Univ., 8050 2nocho, Niigata City, 950-2181, Japan), Yu Murotuka (Niigata Univ., Niigata City, 950-2181, Japan), and Koichi Yoshihisa ((Meijo Univ., Tempaku-ku, Nagoya, 468-8502, Japan)

Acoustic impedance is an important value that determines the boundary condition of each sound field, but collections of actual values are not sufficient for evaluation of many sound fields. First, measurements using a particle velocity sensor for acoustic impedance were tested on various fields. Such measurement results were applied to calculations of sound propagation. Frequency characteristics of sound propagation were obtained along such surfaces as fields of lawns and snow, and areas paved by porous asphalt for drainage. Those characteristics showed fair correspondence with inspected field measurement results. Then, fine calculations in the frequency domain were converted to the impulse response for each sound field model. Convolution operations based on the impulse response and on voice, music, and some noise sources readily produced an ideal sound field to the audible sound file. Furthermore, simulations of the noise, including noise reduction effects from a car running through a drainage pavement area, were executed as advanced applications. Our measurement method of acoustic impedance is inferred to be useful. Calculated sound propagation characteristics based on the measured acoustic impedance correspond to actual propagation characteristics. Audible simulation will be effective as a future means of sound-field evaluation. [Research supported by KAKENHI.]

8:55

5aNS5. Quiet heights and pulse tails: The modal structure of near ground to near ground sound propagation in the nocturnal duct. Roger Waxler (NCPA, Univ. of Mississippi, University, MS 38677, rwax@olemiss.edu)

On clear nights over land a temperature inversion forms, creating an acoustic duct in the first few hundred meters of the atmosphere. Assuming a flat, locally reacting ground surface and a stratified atmosphere, the sound field can be expressed as a superposition of independently propagating vertical modes [J. Acoust. Soc. Am. **112** (6), 2540–2552 (2002) and **115** (4), 1437–1448 (2004)]. The modes account for refraction due to sound-speed gradients as well as for the compliance and resistance of the ground. For typical nocturnal ducts and ground surfaces one finds two types of modes: a surface mode and a set of higher modes. The surface mode is more slowly propagating and more rapidly attenuated than the higher modes. The higher modes' amplitudes all have a sharp minimum several meters above the ground. Two consequences of this modal structure have been observed: At long ranges, narrow-band signals have a range-independent quiet height at which sound levels are reduced by 10 to 15 dB from the levels on the ground [J. Acoust. Soc. Am. **119** (1) 86–95 (2006)] while impulsive signals develop a bandlimited low-frequency tail [J. Acoust. Soc. Am. **88** (1), 455–461 (1990)].

9:15

5aNS6. Influence of short-term variations of meteorological parameters on sound propagation outdoors. Philippe Blanc-Benon, Benjamin Cotte (LMFA, UMR CNRS 5509, Ecole Centrale de Lyon, 69134 Ecully Cedex, France), Benoit Gauvreau, and Michel Berengier (Laboratoire Central des Ponts et Chausses, BP 4129, 44341 Bouguenais Cedex, France)

Predicting long-range sound propagation over a nonurban site with complex propagation media requires the knowledge of micrometeorological fields in the lower part of the atmospheric boundary layer. In the framework of road traffic noise characterization there is a need for reliable sound pressure level predictions for specific propagation conditions that must be representative of time and space (small scale and site effects) characteristics of the acoustic situation. Outdoor measurements involving roughly 100 meteorological and acoustic sensors have been carried out during 3 months in 2005 in the southwest of France. This large database enables us to study the variability of meteorological parameters (wind speed, temperature, etc.) on different time scales. Scaling parameters in the atmospheric surface layer (such as the friction velocity and the Monin-Obukhov length) are estimated every 15 min, which allows looking at relatively short-term variations. Some longer-term effects, from days to months, are studied too. These include occurrences of meteorological configurations and variation of ground parameters. The effect of meteorological variability on sound propagation is studied using parabolic equation simulations. Comparisons between numerical predictions and experimental data will be discussed.

9:35

5aNS7. Effect of atmospheric absorption on aircraft noise propagation in various world regions during the year. Yasuaki Okada, Koichi Yoshihisa (Meijo Univ., 1-501 Shiogamaguti, Tempaku-ku, Nagoya, 468-8502, Japan), and Teruo Iwase (Niigata Univ., Niigata, 950-2102, Japan)

The attenuation of sound due to atmospheric absorption depends strongly on environmental temperature and humidity. For that reason, it varies in response to changing meteorological conditions on a variety of time scales. To perform long-term prediction of outdoor sound propagation more accurately, it is necessary to examine the fluctuation of the attenuation coefficients for atmospheric absorption during the year. Large variations occur according to the time and place in actual meteorological conditions, on which the attenuation coefficients depend strongly. The attenuation coefficients for atmospheric absorption were calculated from hourly meteorological data obtained at ten international airports throughout the world during a year using the calculation method described in ISO 9613-1. Calculated results show that the variability characteristics of the attenuation coefficients during the year depend strongly upon both the sonic frequency and the place. The effects of atmospheric absorption on aircraft noise propagation were examined using the calculated attenuation coefficients. Results showed that the A-weighted sound pressure levels and sound spectra of aircraft noise vary strongly because of the changes in atmospheric absorption, which depend on actual meteorological conditions of various world regions.

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9:55

5aNS8. Acoustic travel-time tomography of temperature and wind velocity fields in the atmosphere. Vladimir E. Ostashev (NOAA, Earth System Res. Lab., Boulder, CO, and Dept. of Phys., New Mexico State Univ., Las Cruces, NM), Sergey N. Vecherin (New Mexico State Univ., Las Cruces, NM), D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Astrid Ziemann, Manuela Barth, and Klaus Arnold (Inst. of Meteorol., Univ. of Leipzig, Germany)

Acoustic tomography of the atmosphere allows one to reconstruct (estimate) temperature and wind velocity fields in the atmosphere and to monitor the evolution of these turbulent fields in time. In the present paper, we report on a progress in construction of a state-of-the-art array for acoustic tomography of the atmosphere, which will allow us to measure travel times of sound propagation between different pairs of sources and receivers within a few meters above the ground. Using these travel times, the turbulent fields will then be reconstructed using different inversion algorithms. The array is being built at the Boulder Atmospheric Observatory, CO, in a collaborative effort between several organizations in the U.S. Furthermore, we discuss several inverse algorithms for estimation of the turbulent fields, including recently developed time-dependent stochastic inversion. Finally, some of these algorithms were used to reconstruct temperature and wind velocity fields in indoor and outdoor tomography experiments carried out by scientists from the Institute of Meteorology, University of Leipzig, Germany. Examples of the reconstructed fields are given. [Work supported by ARO, Grants DAAD19-03-1-0104 and W911NF-06-1-0007.]

10:15–10:30 Break

Contributed Papers

10:30

5aNS9. A comprehensive noise study for the city of Lincoln, Nebraska. Dominique Cheenne, Connie Lee, Michael Cappiello, Sam Lalk, Coleman Martin, and Philip Muzzy (Dept. of Audio Arts & Acoust., Columbia College Chicago, Chicago, IL 60610)

A comprehensive noise study to define the soundscape of the city of Lincoln, NE, has been undertaken by the students enrolled in the Acoustics Program at Columbia College, Chicago. The study includes (1) a global review of best practices in the field of environmental noise control using a perspective emphasizing close attention to effective and proven solutions that can be implemented via policies and effective legislation, (2) the creation of a large-scale outdoor noise propagation computer model that incorporates all primary traffic arterial and readily identifiable noise sources, (3) a comprehensive large-scale testing of noise levels using specially modified noise dosimeters allowing for determination of Leq, and Ldn over 24-h intervals, and (4) a large scale attitudinal survey of the population using a web-based survey engine that allows for the geographical location of the respondents. The results of the survey are used to refine the density of the test locations and the scale resolution of the software model, as well as to provide city and county officials with a targeted course of action pertaining to regulations and noise abatement policies. The project is expected to last 3 years with up to 1200 test locations being analyzed.

10:45

5aNS10. Effect of parameters on modeling low-frequency propagation of petrochemical plant noise over water. Frank Brittain (Bechtel Corp., 2255 Peavine Valley Rd., Reno, NV 89523, fhbritta@bechtel.com) and Xiao Di (Univ. of Mississippi, University, MS 38677)

Modeling noise is vital to designing petrochemical plants to meet community noise limits. Models are used to specify equipment noise, choose add-on controls, and confirm limits will be met. For propagation over land, there are substantial variations of actual levels from the long-term average computed using ISO 9613-2. Commercial modeling software based on ISO 9613-2 excludes propagation over water, where levels are often higher and more variable than at the same distance over land. Over water, higher levels and variability arise primarily from wind, thermal inversions, and lack of attenuations from ground clutter. Often a petrochemical plant and receivers are separated by a large body of water, which reduces modeling accuracy. To better understand propagation over water, full-wave Green's Function-Parabolic Equation (GF-PE) software was used to predict levels between 50 m and 10 km for various lapse rates (temperature increase with elevation) as well as inversion, source, and receiver heights. Plots of attenuation with distance and contours in a vertical plane are presented for 63 Hz. Rates of attenuation considerably less than −6 dB per distance doubling for spherical spreading were found; these and implications for design are discussed. The potential application of GF-PE to verify algorithms for outdoor propagation is indicated.

11:00

5aNS11. Bauxite refinery community noise control and prediction. Jose Augusto Nepomuceno (Acustica & Sonica, R. Fradique Coutinho, 955, sala 01, Sao Paulo, 05416-011 SP, Brasil, janepomuceno@yahoo.com) and Aljan Machado (Alcoa Brasil)

Alumar in the city of Sao Luis in Brazil is one of the largest bauxite processing plants in the world including refinery and reduction plants. The Alumar refinery will expand from approximately 1.4 million mtpy to approximately 3.5 million mtpy and the expansion is expected to be completed in the first half of 2008. Since the year 2000 the authors worked in collaborative way with the in-plant and community noise control and prediction program for the expansion project. The program included noise measurement in the existing plant, development of the Bauxite Refinery Noise Control Manual, several noise control guidelines, and noise modeling to predict community noise impact. The plant is surrounded by different land uses as residential, industrial, rural, and natural permanent protection. Using computer modeling it was studied how the refinery expansion could impact the community areas and what will be the critical equipment requiring special noise limits. The validation process predicted measured results in the range of +2.2 dB(A). The paper presents the noise program results obtained by the year 2006.

11:15

5aNS12. Statistical models for fading and coherence of sound in urban environments. D. Keith Wilson (U.S. Army Engineer Res. and Development. Ctr., 72 Lyme Rd., Hanover, NH 03755, d.keith.wilson@erdc.usace.army.mil), Rafael Bey-Hernandez (U.S. Army Engineer Res. and Dev. Ctr., Hanover, NH 03755), and Vladimir E. Ostashev (NOAA/Earth System Res. Lab., Boulder, CO 80305)

Urban environments typically produce strong, multipath scattering. It is often desirable to characterize the effects of the scattering with statistical models for signal fading and coherence. This paper discusses initial efforts to develop such models. Random configurations of buildings, with varying sizes and number densities, are synthesized. Sound waves are then propagated through the random configurations. The first and second moments of the sound field are calculated by averaging results from the random realizations. Most of the propagation calculations are done by a ray-tracing technique, which is very fast but does not include diffraction. Due to multiple scattering, cases where a receiver is in a full acoustic shadow are infrequent. We also compare the ray tracing to finite-difference, time-domain calculations, which provide a full-wave solution that includes diffraction as well as multiple scattering. Initial results show

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that the wave extinction and coherence diminish exponentially with distance, as is the case for turbulent scattering. The extinction and coherence decay rate increases with the number of buildings and size, but appears to be independent of frequency. We compare the signal probability density functions to statistical fading models developed for radio-wave scattering in urban environments.

11:30

5aNS13. A review on barrier diffraction theories and related insertion loss comparisons in highway noise models. Ning Shu (AZTEC Eng., 4561 East McDowell Rd., Phoenix, AZ 85008), Louis F. Cohn, Roswell A. Harris, and Teak K. Kim (Univ. of Louisville, Louisville, KY 40292)

This paper presents a review on diffraction theories for highway noise barriers. Insertion loss based on different diffraction theories was investigated with two highway noise models, STAMINA and TNM 2.5. STAMINA implements the Kurze and Anderson empirical diffraction formula while TNM 2.5 deploys a simplified MacDonald analytical model. Quantitative analysis indicates that from the perspective of diffraction theory, the average difference of insertion loss between TNM and STAMINA for a point source is about 3 dB(A). To improve the prediction accuracy of TNM, it is recommended that the MacDonald solution, but with real and image sources, should be deployed.

11:45

5aNS14. An acoustic technique for monitoring off-highway vehicle (OHV) use with sound level meter data: Camparison with Trailmaster records. Tina Yack, Ann Bowles (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 95616, tyack@hswri.org), Keith Slauson, William Zielinski (USDA Forest Service, Arcata, CA 95521), and Kenneth Plotkin (Wyle Labs., Arlington, VA 22202)

Off-highway vehicle (OHV) use is of growing interest to land managers. Trail counters are typically used to monitor OHV activity, but accuracy of this technique remains relatively untested. As part of an ongoing study comparing the distribution of American martens (*Martes americana*) with OHV use in the Lake Tahoe Basin, verifiable OHV detections were collected using two techniques, (1) photographs from Trailmaster 1500 camera systems (TMs) placed across roads/trails and (2) continuous 2-s, A-weighted equivalent-continuous sound levels (LAeq2s) collected using Larson-Davis 720 Sound Level Meters (SLMs) placed either near roads/trails or paired with TM animal detection stations. Events were identified in the SLM data by comparing time-history profiles collected in the field with profiles of known snowmobile passes. Errors in SLM event counts were assessed using independent observer logs and compared with the TM records. SLMs were consistently more reliable detectors of snowmobile use. During the Winter 2004 season, TM records were reliably matched to SLM logs only 48% of the time. TMs consistently missed passes and overestimated events compared to SLMs. In conclusion, SLMs proved to be more sensitive and reliable detectors of OHV activity. These results have implications in both wildlife research and land management applications.

SATURDAY MORNING, 2 DECEMBER 2006 WAIANAE ROOM, 7:30 A.M. TO 12:00 NOON

Session 5aPA

Physical Acoustics: Thermoacoustics

Tetsushi Biwa, Cochair
Tohoku Univ., Dept. of Mechanical Systems and Design, Aoba-ku, Sendai 980-8579, Japan

Steven L. Garrett, Cochair
Pennsylvania State Univ., Graduate Program in Acoustics, State College, PA 16804-0030

Contributed Papers

7:30

5aPA1. Amplification of acoustic intensity of a pulse wave propagating through a tube having a temperature gradient. Tetsushi Biwa (Dept. of Mech. Syst. Design, Tohoku Univ., Aoba-ku, Sendai, 980-8579, Japan), Yusuke Tashiro, Masatoshi Shinmei (Nagoya Univ., Chikusa-ku, Nagoya, 464-8603, Japan), and Taichi Yazaki (Aichi Univ. of Education, Igaya-cho, Kariya, Japan)

Ceperley has theoretically pointed out that when acoustic traveling waves pass through a long duct with a positive temperature gradient, the acoustic intensity is amplified as a result of the thermodynamic cycles that the gas undergoes. Even though he only observed damping of the acoustic intensity in the experiment, his idea serves as a starting point of new acoustic devices known as thermoacoustic Stirling heat engines. For a further progress of the thermoacoustic technology, experimental verification of his idea is of great importance. In this paper, we report on the thermoacoustic amplification of acoustic intensity using pulse waves traveling in a long tube equipped with a differentially heated regenerator. A reservoir filled with pressurized air and a solenoid valve were used to generate the acoustic pulse running in one direction in the tube with length 60 m. A series of pressure transducers was used to determine the axial distribution of the acoustic intensity. Use of a pulse wave enabled us to see the frequency dependence of the amplification rate of acoustic intensity very easily. We found that the amplification rate was essentially determined by the ratio of thermal boundary layer thickness to the pore radius of the regenerator.

7:45

5aPA2. Pattern formation in thermoacoustic resonators. German J. de Valcarcel (Dept. de Optica, Universitat de Valencia, Dr. Moliner 50, 46100 Burjassot, Spain), Victor J. Sanchez-Morcillo, and Isabel Perez-Arjona (Universidad Politecnica de Valencia, 46730 Grao de Gandia, Spain)

The phenomenon of pattern formation is ubiquitous in the dynamics of extended nonlinear systems. Patterns arise from the interplay of external forcing, dissipation, nonlinearity, and spatial coupling. These factors are also present in many acoustic systems. However, under usual conditions where the elastic nonlinearity dominates, the development of shock fronts in nonlinear regime strongly limits the possibility of a simple analysis. Viscous media, where thermal nonlinearity dominates over the elastic one and wave distortion is inhibited at short distances, are good candidates for acoustic systems presenting patterns with similar characteristics as those found in other fields, such as hydrodynamics or optics. A model describing

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the evolution of pressure and temperature variations in a resonator with plane walls containing a viscous medium is presented. The large aspect ratio of the resonator allows the transverse spatial coupling, mediated by sound diffraction and temperature diffusion. The homogeneous solutions show bistability at some values of the parameters, in agreement with experimental results reported in the bibliography. We show by means of linear stability analysis that the homogeneous solutions develop spatial instabilities leading to the spontaneous generation of spatial patterns. The analytical predictions are confirmed by a numerical analysis. [Support from Spanish MEC, Project FIS2005-07931-C03-02/03, is acknowledged.]

8:00

5aPA3. Anharmonic acoustic resonators in miniature thermoacoustic cooler. H. El-Gendy, L. Lyard, and O. G. Symko (Dept. of Phys., Univ. of Utah)

In optimizing miniature thermoacoustic coolers for high cooling power density, the acoustic resonator has to be able to sustain very high amplitude acoustic oscillation at the resonant frequency of the driver in order to pump heat up a temperature gradient. Nonlinear effects in cylindrical-shaped resonators lead to acoustic saturation; acoustic energy is dissipated in higher harmonics, thus limiting the intensity of the fundamental mode. To overcome this limitation, noncylindrical resonators were studied here. Conical, exponential, and halfcosine shapes were used. They produce high-amplitude sound waves with essentially no shock waves. Such resonators were chosen here to operate at around 4 kHz, the resonant frequency of the piezoelectric driver. They are easily adapted to the miniature refrigerator. Moreover, they provide an impedance match, which is shape dependent, between the driver and the resonator. Tests at sound intensity levels up to 170 dB show essentially no distortion of the acoustic waveform. Results of their performance will be discussed. [Work supported by ONR and the State of Utah.]

8:15

5aPA4. A study for applying lower temperature heat source to loop-tube thermoacoustic cooling system. Yousuke Imamura, Shin-ichi Sakamoto, and Yoshiaki Watanabe (Faculty of Eng., Doshisha Univ., 610-0321 1-3 Tataramiyakotani Kyotanabe Kyoto, Japan)

To realize the generation of self-sustained sound in lower temperature heat source, a 5-μm in thickness rubber membrane was applied in the loop-tube thermoacoustic cooling system. It is known that the existence of dc flow acts as the negative effect for the loop-tube. The membrane reduces a loss of thermal energy supplied to the prime mover by suppressing the dc flow. So it is considered that lower temperature difference between both the edges of the stack can be realized by using presented system. The locating position of the membrane was moved, and then the temperature differences of the stack were observed. The experiments were carried out on the condition that the thermal energy supplied to the prime mover was constant. It was found that the temperature difference is affected by the position of the membrane. When the membrane was set at the approximate antinode of pressure, the lowest temperature difference was realized. In this condition, the temperature difference with the membrane was 430 K, which was 160 K lower than without the membrane. These results suggest that a lower temperature heat source can be available by setting the membrane on the loop-tube thermoacoustic cooling system.

8:30

5aPA5. Marginal instability of thermoacoustic oscillations of a gas in a tube. Nobumasa Sugimoto and Masatomi Yoshida (Dept. of Mech. Sci., Grad. School of Eng. Sci., Osaka Univ., Toyonaka, Osaka 560-8531, Japan, sugimoto@me.es.osaka-u.ac.jp)

This paper examines a marginal condition of instability of thermoacoustic oscillations of a gas in a tube with one end open and the other closed by a flat wall, subjected to a smooth temperature distribution axially. Assuming a boundary layer is thin compared with the tube radius, the linear theory is developed in the framework of the first-order theory in its thickness. An idea of the method of renormalization enables us to obtain analytical solutions and to derive the marginal condition when the temperature distribution is parabolic. Solving the condition numerically, the marginal curve for the temperature ratio is displayed graphically against the tube radius relative to the boundary-layer thickness. It is found that the temperature ratio has a minimum and that the curve has two branches with respect to the minimum. While the left branch for viscous mode extends to infinity, the right branch, close to the curve for neutral oscillations, asymptotes a certain temperature ratio as the tube radius increases. Such results should be compared with the ones obtained by Rott in the case of a step distribution. Spatial mode of oscillations is also displayed in the marginal state and some discussions on the energy balance are included.

8:45

5aPA6. Direct observation of thermoacoustic energy conversion. Yusuke Tashiro (Dept. of Crystalline Mater. Sci., Nagoya Univ., Furo-cho Nagoya 464-8603, Japan, tashiro@mizu.xtal.nagoya-u.ac.jp), Tetsushi Biwa (Tohoku Univ., Aoba-ku Aramaki Sendai 980-8579, Japan), and Taichi Yazaki (Aichi Univ. of Education, Kariya 448-8542, Japan)

A gas column starts to oscillate when the externally imposed temperature gradient exceeds a critical value. Also, when an acoustic wave propagates through a differentially heated regenerator, the acoustic intensity is thermally amplified. Such thermoacoustic phenomena attract considerable interest not only because it becomes possible to develop pistonless Stirling heat engines, but because an alternative approach to understand a heat engine has been proposed based on energy flows. In this work, we report the direct observation of the thermoacoustic energy conversion in a tube filled with atmospheric air to test the validity of the proposal. A gas column was partly heated by an electrical heater and driven by an oscillating piston at 1.00 Hz to assure good thermal contact with the surrounding tube wall. We simultaneously measured pressure and velocity oscillations to determine the acoustic intensity. As a result, we found the amplification of acoustic intensity in the region with a positive temperature gradient. This supports the proposed approach. Further test was made by comparing the acoustic power produced per unit length and the thermodynamic cycles of the gas, the latter of which was also experimentally determined from measurements of temperature and pressure.

9:00

5aPA7. Study on a thermoacoustic cooling system to drive the fundamental resonance frequency by connecting a triggered tube. Hideo Yoshida, Shin-ichi Sakamoto, and Yoshiaki Watanabe (Ultrasonic Labs., Doshisha Univ., 1-3 Miyakodani Tatara, Kyotanabe City, Kyoto 610-0321 Japan, dtf0166@mail4.doshisha.ac.jp)

A thermoacoustic cooling system to drive a looped-tube with the fundamental resonance frequency by connecting a triggered tube is proposed. The heat pump is located in the triggered-tube. The existence of dc flows, which is caused by harmonics in the looped tube, has been pointed out as negative effects. The presented thermoacoustic cooling system, however, connected the triggered-tube is expected to suppress the harmonics and to improve cooling effect, so that the length of the triggered tube is designed to have the fundamental resonance frequency. It is well known that the phase relation between sound pressure and particle velocity is important, and approximately in-phase relation is the best position for cooling effect. In the presented system, the sound energy is supplied from the looped tube. It is important to know the best connecting position where the phase relation is approximately in phase. The sound pressures were observed, then the phase relations were calculated by two-sensor power method. By using results, the best position of the triggered tube was decided. It is confirmed that the presented system can realize the suppression of harmonics and the decrease of outlet temperature. It is also confirmed that the decrease of temperature has the higher effects compared with the looped tube in the same environmental conditions.

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9:15

5aPA8. Determination of complex propagation constant, acoustic intensity, and acoustic power in an arbitrarily terminated pipe using laser Doppler anemometry.

Ki Won Jung and Anthony A. Atchley (Grad. Program in Acoust., Penn State Univ., University Park, PA 16802, kuj102@psu.edu)

Microphones are widely used to determine acoustical parameters, such as the complex propagation constant and acoustic intensity and power, of sound fields. It is less common to use measurements of acoustic particle velocity for the same purpose. Precise measurement of acoustic particle velocity can be achieved using laser Doppler anemometry (LDA). Although this measurement technique requires optical access to the region of interest and the use of seeding particles, in some circumstances it can be less invasive and offer greater flexibility than measuring pressure fields with microphones. The focus of this research is to use LDA to determine the complex propagation constant, acoustic intensity, and power inside a constant cross-section circular pipe. The measured particle velocities are fit to a counterpropagating plane-wave model to determine the complex amplitudes and propagation constant. Other acoustic quantities, such as the radial-dependent acoustic intensity and the cross-sectional-averaged acoustic power, can then be calculated. The complex propagation constant is compared with the theoretical value based on thermoviscous boundary layer theory. The acoustic intensity and power are compared with numerical solutions to Rott's wave equation. [Work supported by the Penn State Graduate Program in Acoustics.]

9:30

5aPA9. Spectral gaps and discrete transmission in slender tubes.

Manvir S. Kushwaha (Inst. of Phys., Univ. of Puebla, P.O. Box J-45, Puebla, Pue. 72570, Mexico), A. Akjouj, B. Djafari-Rouhani, L. Dobrzynski, and J. Vasseur (Univ. of Sci. and Technol. of Lille-I, France)

Extensive band structure and transmission spectra for the longitudinal (acoustic) wave propagation in a system made up of N' dangling side branches (DSBs) periodically grafted at each of the N equidistant sites on a slender tube are reported. A periodic pattern of large stop bands is obtained for the airy DSB on a slender tube. The emphasis is laid on the interesting result of huge gaps and discrete transmission spectrum due only to the DSB grafted at a *single* site ($N=1$) on the slender tube. Designing the system with open tubes allows achievement of lowest gap below a threshold frequency and extending up to zero—thereby providing an entirely discrete band structure and transmission spectrum. This should have important consequences for the suppression of low-frequency noise and for designing filters and transducers.

9:45

5aPA10. The effects of turbulence on an undular bore.

Pablo Luis Rendon (CCADET, Universidad Nacional Autonoma de Mexico, Ciudad Universitaria, Mexico D.F. 04510, Mexico, rendon@aleph.cinstrum.unam.mx)

The Korteweg-de Vries-Burgers (KVB) equation is the simplest equation to incorporate simultaneously the effects of nonlinearity, damping, and dispersion, and is of the form $\eta_t + \eta, \eta_x + \eta_{xxx} = \delta, \eta_{xx}$. It has been found that the steady solution to this equation exhibits all the characteristics of the undular bore, with $\eta(x,t)$ the surface profile. An examination of the effects of turbulence on undular bores, and of whether this turbulence can effectively alter the overall shape of the bore, is conducted through numerical simulation of the KVB equation. [The author thanks the Universidad Nacional Autonoma de Mexico for funding this project through a PAPIIT (IN116205) grant.]

10:00–10:15 Break

10:15

5aPA11. Resonant aero-acoustic excitation of cavity depth modes.

Boris M. Efimtsov, Alexey Yu. Golubev (TsAGI, 17 Radio St., 107005 Moscow, Russia), and Anders O. Andersson (Boeing MC 67-ML, Seattle, WA 98118)

Results are generalized of parametric experimental investigations of resonant aero-acoustic excitation of the depth modes of a flow-grazed cavity at low-subsonic (U50 m/s) and transonic flows (0.7M1.045). Special attention is given to the fundamental mechanisms responsible for this phenomenon. The conditions for these mechanisms to manifest themselves and interact are predicted. Observations are made of the relations between the characteristic flow and cavity parameters at which the maximum resonant aero-acoustic response of the cavity depth mode takes place. These relations take into account the distinction between the flow velocity and the characteristic spreading velocity of the disturbances downstream of the wall discontinuity as well as the distinction between the cavity depth and the characteristic linear scale in the Helmholtz number connected with the edge effect for sound waves in the vicinity of the cavity gap. Experimental data illustrating the reliability of these relations are presented. The extent of the Strouhal-number region where the cavity mode excitation can be treated as an aero-acoustic resonance is determined.

10:30

5aPA12. Single transducer parametric sound source.

Bastian Epp and Volker Mellert (Univ. of Oldenburg, Faculty V, Inst. of Phys., D-26111 Oldenburg, Germany)

The idea of using the nonlinear effects of the propagation of sound waves with finite amplitude to build up highly directed sound sources is well known. In underwater acoustics, such parametric arrays are widely used. There are also attempts to build up parametric arrays in air to generate highly directed air-borne sound. To be able to use nonlinear effects in air as a carrier medium, one needs high amplitudes of the propagating sound wave. Consequently, a superposition of single sound sources (arrays) is often reported in the literature to increase the size, and so the directivity, of the sound source as well as the acheived sound pressure level. A major disadvantage of using arrays is the problem of slightly varying resonance frequencies and phase characteristics of the single source components as well as a complicated superposition of single directivity patterns. It is difficult to investigate basic properties of wave propagation and the nature of distortion in such a way. In this work, a single, high-power ultrasound transducer was used to radiate sound waves with finite amplitude, able to generate extraneous frequency components in air. Using only a single transducer, it was possible to investigate basic properties of nonlinear wave propagation in air.

10:45

5aPA13. Scale model experiments over curved rough surfaces.

James Chambers and Andrew Whelan (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677)

Previous research has indicated that the propagation of sound over a curved surface can be drastically altered by the presence of continuous small scale roughness that is smaller than the acoustic wavelength. This effect can alter sound levels in the shadow zone from $+6$ to -20 dB relative to levels past a similar smooth surface depending on the frequency of interest. These findings were originally observed for a single radius of curvature and a limited range of roughness sizes. The work has been revisited for a larger range of radii of curvature and roughness sizes with comparable results. The results for this problem as well as the related one of propagation in an upwardly refracting medium will be discussed. [Work supported by the Army Research Laboratory.]

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11:00

5aPA14. Low-frequency pulse propagation over irregular terrain. Xiao Di and Kenneth E. Gilbert (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

The shape of a low-frequency pulse that propagates to the far side of a hill depends primarily on two physical mechanisms: a diffracted wave and a surface wave. Both of these mechanisms depend on frequency, so that the pulse shape and frequency spectrum of the received pulse differ significantly from those near the source. As is well known, diffraction acts like a low-pass filter. The surface wave has a more complicated frequency dependence that depends on the terrain, the frequency dependence of the ground impedance, and the near-ground sound speed profile. For a given range, these parameters determine a frequency band that contains most of the surface wave energy. The surface wave thus functions as a low-frequency bandpass filter. The combined effects of diffraction and surface wave propagation on pulse propagation over a hill are studied using a parabolic equation model and a piecewise linear mapping to flatten the irregular terrain. Numerical calculations are presented for selected terrain features, ground impedances, and sound speed profiles. [Research supported by the U.S. Army Armament, Research Development and Engineering Center (ARDEC).]

11:15

5aPA15. Proper orthogonal decomposition for sensitivity analysis and classification of outdoor sound propagation. Chris Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., 590 Holloway Rd., MS 11-B, Annapolis, MD 21402, pettitcl@usna.edu) and D. Keith Wilson (U.S. Army Cold Regions Res. and Eng. Lab., Hanover, NH 03755)

Simulations of outdoor sound propagation are compromised by many sources of uncertainty and error. A key step in validating computational acoustics models, which may be interpreted as the act of assessing predictive skill, is estimating ensemble sound pressure variability due to uncertainties in the parameters that define the propagation conditions. This paper describes elementary steps toward achieving this goal. The atmospheric surface layer is represented through Monin-Obukhov similarity theory and the acoustic ground properties with a relaxation model. Randomness is assumed across appropriate ranges of the governing parameters. The parameters are modeled herein as independent random variables, but future efforts will extend this formulation to include random field models of the parameters with varying levels of correlation. Sound propagation is predicted with the parabolic equation method. Latin hypercube sampling (LHS) and proper orthogonal decomposition (POD) are employed to develop low-dimension representations, i.e., reduced order models, of the sound pressure random field. Sensitivity of the sound pressure field is studied via the model-induced sensitivity of the POD mode coefficients to the system parameters as well as statistical dependence on the number of LH samples. Attention is also given to connecting the POD coefficients to documented indicators of refraction conditions.

11:30

5aPA16. Analysis of the effect of small currents on ray stability and caustic formation in a layered moving fluid medium. David Bergman (Lockheed Martin Maritime Systems and Sensors, 199 Borton Landing Rd., Moorestown, NJ 08570)

In studies of underwater sound it is usually sufficient to treat the medium of propagation as a static inhomogeneous medium. Including current and time-dependent changes adds to the overall understanding of the behavior of sound in underwater environments. The quiescent medium approach combines the movement of the fluid with the local sound speed to create an effective sound speed profile. Recently it has been demonstrated that the effects of small-scale fluid motion on the propagation of sound in the ocean can be of the same order of magnitude as that of the static sound speed profile [J. A. Colosi, J. Acoust. Soc. Am. **119**, 705–708 (2006)]. The stability of the acoustic ray equation in a moving fluid medium depends on the first and second derivatives of the sound speed profile and fluid velocity [D. R. Bergman, Waves Random Complex Media **15**(4), 417–435 (2005)]. In this presentation the effects of weak but rapidly changing currents on the stability and intensity of the acoustic field are investigated for environments with low Mach number. The main result will show that in real underwater environments small currents cannot be ignored when modeling or analyzing the propagation of sound.

11:45

5aPA17. Complex drag coefficient and complex Nusselt number for oscillatory flow. Jin Liu (United Technologies Res. Ctr., East Hartford, CT 06108) and Steven L. Garrett (Penn State Univ., State College, PA 16804)

Characterizing and choosing the optimal porous material for efficient heat transfer while minimizing viscous loss is important for improving the efficiency of thermoacoustic machines. The pressure drop for oscillating flow through porous media has both a viscous (real) part and an inertial (imaginary) part, making the drag coefficient for oscillatory flow a complex number [J. Liu, J. Acoust. Soc. Am. **115**, 2380 (2004)]. Similarly, a phase difference exists between the oscillating pressure and the oscillating temperature. A complex Nusselt number is therefore required to characterize heat transfer for oscillating pressure. For the first time, the complex Nusselt number will be related to the thermoviscous f-function favored by thermoacousticians. This identification also allows the heat transfer coefficient for oscillating flow to be determined experimentally, at sufficiently low amplitudes, without requiring the measurement of temperatures or the use of heat exchangers. At low frequencies, the real part of the complex Nusselt number converges to the steady-state (dc) heat transfer coefficient. At high frequencies, it recovers the results published by other researchers interested in the behavior of heat exchangers used in oscillating flows. [Work supported by the ONR.]

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Session 5aPP

Psychological and Physiological Acoustics: Miscellaneous: Pipe Organs to T.V. Ads

Peter Zhang, Cochair
Michigan State Univ., Dept. of Physics and Astronomy, East Lansing, MI 48824

Minoru Tsuzaki, Cochair
Kyoto City Univ. of Arts, Faculty of Music, 13-6 Kutsukake-cho, Oe, Nishikyo-ku, Kyoto 610-1197, Japan

Chair's Introduction—8:30

Contributed Papers

8:35

5aPP1. The significance of relative loudness in listener identification of pipe organ stops. Alastair C. Disley and David M. Howard (Dept. of Electron., Univ. of York, Heslington, York, YO10 5DD, UK)

As part of a wider study of the factors affecting listener identification of pipe organ stops, the authors' previous research [A. C. Disley and D. M. Howard, "Onset transient significance in listener identification of pipe organ stops," J. Acoust. Soc. Am. **119**, 3333, abstract only (2006)] has suggested that the relative loudness of a stop as presented to a listener may have a significant effect on its identification. Recordings of a representative selection of pipe organ stops were played to a group of 43 organists in a formal listening test. Each stop was presented twice, once at its original amplitude and once at an increased amplitude equal for all stops. Listeners were asked to classify each example using nine general categories of organ stops. The results suggest that increased amplitude improves listener recognition for some stops (particularly open flute stops), but produces significant misidentification of other stops, especially those with the greatest difference between the original and increased amplitude. These results suggest that caution should be employed when considering equalizing sample amplitudes in similar listening experiments.

8:50

5aPP2. Influence of attack transient and decay times of percussive sounds on the echo threshold. Hari V. Savitala and Jonas Braasch (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, hari.savitala@gmail.com)

The human ability to localize a direct sound source in the presence of reflected sounds is well known as the precedence effect. The echo threshold defines the time delay of the reflection, above which the reflection becomes audible as a separate event. For musical instruments, various attack transient and decay times affect the echo threshold, but often a constant echo threshold of 50 ms is assumed for music. The aim of this study is to determine in more detail how the parameters of a test sound affect the echo threshold. For this purpose, 12 adult listeners with normal hearing participated in a psychoacoustic experiment to determine the echo threshold for various percussive and nonpercussive instrument sounds. In the headphone-based experiment, the lead was placed with an interaural time difference (ITD) of ± 300 μs, while the lag was presented from the opposite side. Interstimulus interval (ISI) values of 0, 1, 2, 5, 10, 25, 50, and 100 ms were tested. The results of the experiment showed a large variation of the echo threshold. The smallest value of 7.50 ms was found for the castanet and the largest of 75.00 ms for a free-reed organ pipe.

9:05

5aPP3. Remembering what's missing: Modeling phoneme restoration and other complex forms of auditory induction as the triggering of a memory. Justin Aronoff (Neurosci. Program, Univ. of Southern California, HNB 27, Los Angeles, CA 90089, aronoff@usc.edu)

Although phoneme restoration is often spoken about and modeled as a unique phenomenon specific to language, the effect is very similar to restoration effects found with well-learned musical melodies. This suggests that both of these effects may be the result of a natural characteristic of learned temporal patterns, rather than a special mechanism. A recurrent connectionist network was designed to test this hypothesis. The network contained ten input and output nodes, which represented notes in a melody. The model was trained on sets of simple melodies, each consisting of eight randomly selected sequential notes. To test that the model was learning and identifying the melodies, the network was trained to predict the next note in the tune, which could only be accomplished by maintaining a representation of the sequence of prior notes. After the network learned the melodies, the fourth sequential note in each tune was removed and replaced by activation across all the input nodes, simulating noise. The model demonstrated accurate restoration, performing significantly better than chance, both at the noise position and over the notes that followed, suggesting that sequence learning may be sufficient to explain phoneme restoration and other forms of complex auditory induction.

9:20

5aPP4. Auditory priming releases Chinese speech from informational masking. Zhigang Yang, Jing Chen, Xihong Wu, Yanhong Wu (Dept. of Psych., Natl. Key Lab. on Machine Percept., Peking Univ., Beijing, China, 100871), Bruce Schneider, and Liang Li (Univ. of Toronto at Mississauga, Mississauga, ON, Canada L5L 1C6)

Before an English speech sentence is presented, hearing or reading the sentence without the last key word improves recognition of the last key word if the full-length speech sentence is presented under speech masking but not under noise masking. This phenomenon suggests a content priming effect on releasing speech from informational masking. To determine whether the priming effect extends to tonal Chinese speech, and, in particular, whether it can be induced by the target talkers voice, in the present study, listeners were presented with either same-voice/different-sentence primes or same-voice/same-sentence primes before hearing the target sentence in either two-talker-speech masking or noise masking. Under speech masking, each of the two prime types significantly improved recognition of the last key word in the full-length target sentence, but the content priming is stronger than the voice priming. Under noise masking, same-voice/same-sentence primes had a weak but significant priming effect, but same-voice/different-sentence primes had only a negligible priming effect. These results suggest that both content and voice cues can be used by listeners to release Chinese speech from informational masking, but only

5a SAT. AM

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content cues are useful for releasing Chinese speech from energetic masking. [Work supported by China NSF and Canadian IHR.]

9:35

5aPP5. Auditory memory for frequency in sequential interference. Jeffrey J. DiGiovanni and Dennis T. Ries (School of Hearing Speech and Lang. Sci., Ohio Univ., Athens, OH 45701)

Percepts within working memory are weakened by the presence of intervening stimuli that lie along the same perceptual dimension. To this end, the effect of sequential interference on the storage of frequency information within auditory working memory was measured. The difference limen for frequency was measured by the method of constant stimuli for two diotic and two dichotic conditions. The first diotic condition served as a control and consisted of two 300-ms stimuli separated by 5 s. The frequency of the referent (first tone) was roved over a 100-Hz range centered at 435 Hz. Listeners indicated whether the target (second tone) was higher or lower in frequency than the referent. The second diotic condition added four interfering tones between the referent and the target. In the third condition, the interferers and referent/target pair were lateralized to opposite ears via the Stenger effect. In condition four, the phase of the interferers differed between ears by 180 deg. Results indicate (1) the addition of interferers increased the difference limen for frequency, and (2) when the interferer and target/referent pair differed along a single localization dimension, the effect of the interferers was reduced.

9:50

5aPP6. Speech recognition in coherently-amplitude modulated noise. Imran Dhamani, K. Naveen, and B. Rajashekhar (Dept. of Speech and Hearing, MCOAHS, MAHE, Manipal, Karnataka, India, imrandhamani@yahoo.co.in)

Comodulation masking release is a phenomenon that improves the detectability of a masked pure tone or speech signal by addition of a coherently amplitude-modulated energy above and/or the signal frequency. The majority of the studies done on comodulation masking release have studied threshold detection for a pure tone, or speech identification at or near threshold levels with a favorable signal-to-noise ratio. Moreover, various studies examining comodulation masking release under conditions of reduced frequency selectivity in listeners with cochlear hearing impairment indicate absence or reduced comodulation of masking release effect (Hall *et al.*, 1984; Hall and Grose, 1985; Moore *et al.*, 1993). In the current study, the speech recognition task was studied for the CMR effect at supra-threshold level and unfavorable signal-to-noise ratio of -10 dB. Results indicated absent CMR effect at supra-threshold level for the word recognition task for normal-hearing individuals.

10:05–10:20 Break

10:20

5aPP7. Modulation detection thresholds for cochlear implants: Dependence on stimulation site and stimulus level. Bryan E. Pfingst, Rose A. Burkholder, Catherine S. Thompson (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI 48109-0506, bpfingst@umich.edu), and Li Xu (Ohio Univ., Athens, OH 45701)

Amplitude modulation of pulse trains is the strategy of choice for transmission of temporal-envelope information in most modern auditory prostheses. Previous studies by Fu, Colletti, and Shannon have shown high negative correlations between modulation detection thresholds (MDTs) (measured at only one site in the electrode array) and speech recognition with cochlear or brainstem auditory prostheses. The objective of this study was to characterize MDTs across the parameter space used in cochlear prosthesis speech processors: all usable stimulation sites at multiple stimulus levels. Across-site patterns of MDTs varied considerably from subject to subject. In general, sites with the poorest MDTs were located in the basal third of the scala-tympani electrode array, but this pattern did not hold for all subjects or all stimulus conditions. The mean MDT for two levels (30% and 70% of dynamic range) was highly correlated with the mean for five levels. Mean MDTs at these two levels averaged over four adjacent electrodes in any of five segments of the electrode array were negatively correlated with speech recognition scores. This study suggests that in order to adequately characterize a subject's modulation detection acuity, it is necessary to sample MDTs at multiple sites and levels. [Work supported by NIH/NIDCD.]

10:35

5aPP8. Lateralization of sine tones—Interaural time versus interaural phase. Peter X. Zhang and William M. Hartmann (Dept. of Phys. and Astron., Michigan State Univ., 4230 BPS Bldg., East Lansing, MI 48824)

Five listeners estimated the lateral positions of 50 sine tones in a headphone experiment designed to determine whether the human sense of sound location correlates better with the interaural phase difference (IPD) or the interaural time difference (ITD). In any experimental block the IPD values ranged from -150 to $+150$ degrees, and the frequencies were chosen such that the ITDs ranged from -1000 to $+1000$ microseconds. The frequencies were all in the range where human listeners are known to be able to lateralize tones based on the ITD in the waveform fine structure. It was found that the lateralization responses correlated with the ITD much better than with the IPD. The average variance was five times smaller for the ITD hypothesis compared to the IPD hypothesis, and only the ITD hypothesis led to a well-fitting (compressive) function. For the ITD function, individual compressive exponents varied considerably and averaged 0.75. For the IPD function the exponents were too small to be meaningful. Comparison with previous lateralization studies indicates the importance of presenting the entire range of stimulus parameters in all experimental blocks because it appears that listeners tend to use the entire range of allowable responses in any block. [Work supported by the NIDCD.]

10:50

5aPP9. Influence of visual stimuli in the sound quality evaluation of loudspeaker systems. Alex Karandreas and Flemming Christensen (Sound Quality Res. Unit, Dept. of Acoust., Aalborg Univ., Fredrik Bajers Vej 7 B5, DK-9220 Aalborg, Denmark)

There is currently an increasing demand to evaluate sound quality attributes of products and to understand to what extent they influence a user's overall impression, since there is usually more than one modality stimulating this evaluation. The present study uses loudspeakers as an example and evaluates the overall impression in relation to hearing and vision. In order to quantify the bias that the image of a loudspeaker has on the sound quality evaluation done by a naive listening panel, loudspeaker sounds of varied degradation are coupled with positively or negatively biasing visual input of real loudspeakers, and in a separate experiment by pictures of the same loudspeakers. In order to choose loudspeakers that provide a sufficient range of visual bias a preliminary visual-only experiment has been conducted. From the ongoing experiments it is possible to evaluate how much the auditory perception of a loudspeaker can be biased from visual input and study how the two modalities interact. Results from the experiments are presented.

11:05

5aPP10. Head-related transfer functions via the fast multipole accelerated boundary element method. Nail A. Gumerov, Ramani Duraiswami, and Dmitry N. Zotkin (Perceptual Interfaces and Reality Lab., Inst. for Adv. Comput. Studies, Univ. of Maryland, College Park, MD 20742)

Computation of head-related transfer functions (HRTF) via geometrical meshes and numerical methods has been suggested by a number of authors. An issue facing this approach is the large computational time needed for high frequencies, where the discretization must include hundreds of thousands of elements. Conventional computational methods are unable to achieve such computations without excessive time or memory. We use a newly developed fast multipole accelerated boundary element

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method (FMM/BEM) that scales linearly both in time and memory. The method is applied to the mesh of the widely used KEMAR manikin and its HRTF computed up to 20 kHz. The results are compared with available experimental measurements.

11:20

5aPP11. Perceptual measurement of sound level differences between TV programs and advertisements. Eiichi Miyasaka (Musashi Inst. of Technol., 3-3-1, Ushikubo-Nishi, Tsuzuki-ku, Yokohama, Japan) and Takahiro Kamada (Pioneer Co., Tsurugashima, Saitama, Japan)

In order to investigate loudness difference between TV program materials and advertisements (CMs) inserted in the materials, sound levels of CMs broadcast in five terrestrial broadcasting stations throughout a day were quantitatively measured and a perceptual experiment was performed. The averaged sound levels (ASLs) and the standard deviations (SDs) of 4262 CMs were concentrated on −7 dB and 5 dB, respectively. Next, three types of CMs with different ASLs and SDs were used to the perceptual experiment. These ASLs and SDs were (−5.1 and 1.8 dB for CM-1), (−7.1 and 4.9 dB for CM-2), (−14.2 and 10.1 dB for CM-3), respectively. The reference sound as a main program material was speech uttered by a NHK female announcer with duration of 10 s. The ASL and the SD were −6.5 and 10.1 dB, respectively. The transformed up-down method was used for 12 observers with normal hearing. The results show that all CMs were perceived at least 3 to 5 dB louder than the reference for a half of the observers. These results show that there is large perceptual discrepancy between CMs and the main program materials. [Work supported by The Okawa Foundation.]

SATURDAY MORNING, 2 DECEMBER 2006 HONOLULU ROOM, 8:30 TO 11:30 A.M.

Session 5aSA

Structural Acoustics and Vibration: Numerical Methods in Structural Acoustics and Vibration

Dean E. Capone, Cochair

Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804

Toshimitsu Tanaka, Cochair

Kobe Steel, Ltd., Mechanical Engineering Research Lab., 1-5-5 Takatsukadai, Nishi-Ku, Kobe City, Hyogo 651-2271, Japan

Invited Papers

8:30

5aSA1. Recent developments and applications of energy finite- element analysis. Kuangcheng Wu (Dept. of Signatures and Hydrodynamics, Northrop Grumman Newport News, Newport News, VA 23607) and Nickolas Vlahopoulos (Univ. of Michigan, Ann Arbor, MI 48109)

The energy finite-element analysis (EFEA) comprises a finite-element-based solution for vibration and acoustic modeling that uses energy-based variables for formulating the governing differential equations. This presentation covers the main points in the derivation of energy finite-element analysis (EFEA), validation through comparison with test data, and case studies that demonstrate the computational capabilities of the EFEA. The EFEA can be applied in various engineering disciplines as it will be demonstrated by the case studies. Thus, it comprises a general purpose simulation method for vibration and acoustic analysis of complex systems. [Work support for this research is provided by ONR code 331.]

8:50

5aSA2. Fast finite-element analysis for damping of automotive structures having elastic bodies, viscoelastic bodies, porous media, and gas. Takao Yamaguchi (Dept. of Mech. System Eng., Gunma Univ., 1-5-1 Tenjin-cho, Kiryu, Gunma 376-8515, Japan) and Yoshio Kurosawa (Fuji Heavy Industries, ltd., Sano-shi, Gunma, 327-0512 Japan)

A numerical method is proposed to calculate damping properties for automotive soundproof structures involving solid bodies, porous media, and air in two-dimensional regions. Both effective density and bulk modulus have a complex quantity to represent damped sound fields in the porous media. Particle displacements in the media are discretized using a finite-element method. For damped solid bodies, displacements are formulated using conventional finite elements including complex modulus of elasticity. Displacement vectors as common unknown variables are solved under coupled condition between solid bodies, porous media, and gas. Further, by applying an asymptotic method to a complex eigenvalue problem, explicit expressions of modal loss factor for the mixed structures are derived. The proposed methods yield appropriate results for some typical problems and this method diminishes computational time for large-scaled finite-element models. Moreover, it is found that damping can be coupled in the mixed structures. An expression to calculate a share of dissipated energy for each element in mixed structures is also derived. Damping behaviors in sound bridge phenomena are analyzed using the proposed method.

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9:10

5aSA3. Vibro-acoustic analysis using hybrid finite element, boundary element, and statistical energy analysis models. Phil Shorter and Vincent Cotoni (ESI US R&D, 12555 High Bluff Dr., Ste. 250, San Diego, CA 92130, pjs@esi-group-na.com)

This paper presents an overview of the theory and application of the hybrid FE-SEA method. The method provides a rigorous way to model the fully coupled response of a vibro-acoustic system across a broad frequency range using a combination of FE, BEM, and SEA. The theory of the method will be discussed and results from the following applications will be presented: (i) modeling the transmission loss and radiation efficiency of trimmed automotive subassemblies; (ii) creating efficient models of structure-borne noise in an automotive body-in-white; (iii) modeling structure-borne noise transmission in a commercial aircraft fuselage; and (iv) modeling the response of a satellite antenna to broadband acoustic loading.

Contributed Papers

9:30

5aSA4. Simulation of borehole acoustic measurements with adaptive finite elements for the accurate and efficient assessment of rock formation properties. Christian Michler, Leszek Demkowicz, and Carlos Torres-Verdin (Inst. for Comput. Eng. and Sci., Univ. of Texas, 201 East 24th St., Austin, TX 78712, c.michler@ices.utexas.edu)

Borehole acoustic measurements are routinely used to probe *in situ* properties of rock formations. The numerical simulation of such measurements involves numerous outstanding challenges, such as the modeling of large spatial variations of rock porosity and anisotropy, material diffractions, source-sensor effects, and the need for a reflectionless truncation of the computational domain. To simulate borehole acoustic measurements, a new axisymmetric finite-element formulation in the frequency domain has been developed and successfully tested. The formulation couples acoustic phenomena within the fluid-filled borehole with elasticity in both the probed rock formations and the measurement tool. One of the central components of our formulation is the automatic adaptivity of element size and approximation order of the finite-element spatial discretization. No user interaction is necessary to control the adaptivity. Such a formulation enables accurate simulation results in the presence of large contrasts of material properties. A novel combination of the perfectly matched layer enhanced with finite-element adaptivity effectively truncates the computational domain. Several challenging benchmark problems are presented that confirm the accuracy, reliability, and efficiency of our method to reproduce complex waveforms excited by monopole and dipole sources in a fluid-filled borehole that penetrates layered media.

9:45

5aSA5. Perfectly matched layers for the numerical approximation of the radiation boundary condition in structural acoustic finite element tools. Mario Zampolli, Alessandra Tesei, Finn B. Jensen (NATO Undersea Res. Ctr., La Spezia, Italy), and John B. BlottmanIII (Naval Undersea Warfare Ctr., Newport, RI 02841)

The Berenger perfectly matched layer (PML) is one of a variety of numerical techniques for approximating the Sommerfeld radiation boundary condition. In acoustics, the PML is a nonphysical finite thickness layer of fluid or solid material that surrounds the physical computational domain of interest and acts as an anisotropic absorber of the outgoing waves. The absorption coefficients in the PML are such that the acoustic energy is dissipated selectively only in the direction perpendicular to the interface between the PML and the physical domain. The study presented here shows that the appropriate choice of PML absorbing functions can have a beneficial effect on the convergence of frequency domain acoustic finite-element tools. In the context of elastic target scattering computations, it is shown how the PML can be applied in direct contact with the wet surface of convex elastic targets. Other examples from low-frequency geoacoustic benchmarking applications demonstrate the adaptability of the PML to a variety of geometries and to problems where the radiation condition must be imposed across inhomogenous layers. The numerical results obtained with the finite-element/PML tool are compared to results from finite-element/infinite-element codes and to results from parabolic equation models.

10:00–10:15 Break

10:15

5aSA6. Reconstruction of vibro-acoustic response of a plate using Helmholtz equation least-squares method. Huancai Lu and Sean F. Wu (Mech. Eng. Dept., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, huancai.lu@gmail.com)

A numerical investigation of reconstructing the vibro-acoustic responses of an arbitrary structure subject to vibration excitations based on the Helmholtz equation least-squares (HELS) method is presented. It is emphasized that in many engineering applications, the exact solution to a general vibrating structure does not exist, and the HELS method is one way of getting approximate solutions in a cost-effective manner. In this study, the test object is a simply supported, unbaffled thin plate. The reason for selecting this simply supported plate is that the analytic solutions to the plate vibrations are readily available. The field acoustic pressures generated by the Rayleigh integral are taken as input to HELS algorithms to reconstruct the normal velocity and normal acoustic intensity on the plate surface using Tikhonov regularization associated with generalized cross-validation methods. The reconstructed normal surface velocities are compared with the benchmark values, and the out-of-plane vibration patterns at the first five natural frequencies are compared with natural modes of the simply supported plate. The effects of measurement aperture, standoff distance, and location of the origin of the coordinate system on the resultant accuracy of reconstruction are examined. [Work supported by NSF.]

10:30

5aSA7. Application of the short-time Fourier transform and the two-dimensional Fourier transforms to detect damage in plates. Jamal Assaad, Faysal El Youbi, Sebastien Grondel, and Emmanuel Moulin (IEMN-DOAE, 59313 Valenciennes, cedex 9, France)

Lamb waves are widely used in the fields of characterization and nondestructive testing of structures. Usually, the full characterization of the structures requires two kinds of information: the first one is related to the material quality, the other one aims at the evaluation of the structure health, i.e., the detection of localized damage or defects. Time-frequency analyses or two-dimensional Fourier transforms (2DFT) have been used separately to solve such a problem. The purpose in this work is therefore to present a dual signal processing approach based on the relationship between the STFT (short-time Fourier transform) and the 2DFT amplitudes that allows a better interpretation and identification of Lamb modes. This relationship will be numerically verified by a two-dimensional finite-element method. Experimentally, a 3-mm-thick aluminum plate with an emitter and a receiver are considered. The emitter consists of two piezoelectric transducers while the receiver consists of a 32 multi-electrode piezoelectric transducer. This receiver has been developed especially to increase signal-to-noise ratio at the reception. In order to simulate damages a hole with different diameter was introduced in the plate center. The application of this dual signal processing has located conversion modes phenomenon.

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10:45

5aSA8. Sound radiation from finite plates excited by a space-time stream of random impulses. Katsuji Akamatsu (Machinery Acoust., 1-1-2-314 Obanoyama Shinohara, Nada-ku, Kobe 657-0015, Japan, akamatsu@s4.dion.ne.jp), Takao Yamaguchi (Gunma Univ., Kiryu 376-8515, Japan), and Junichi Kanazawa (Musashino Damping Technol., Tachikawa 190-0002, Japan)

Sound radiation from baffled finite elastic plates that are subjected to impulses occurring at random points on the surface, at random intervals, and with random strength is analyzed. The plate response is obtained in the Stieltjes integral form of the plate impulse response function with respect to the occurrence of impulses. The total sound power radiated from the plate is formulated according to Heckl's approach. Explicit expressions for the expected value of the plate response and the radiated sound power are derived for the case in which the stream of impulses is uncorrelated. An approximate solution for the radiated sound power is obtained by assuming light damping of plates and by neglecting modal coupling effects. For comparison, the exact and approximate solutions are evaluated numerically for a plate with constant loss factors. The analysis is applied to the prediction of rainfall noise by expressing the expected value of the exciting force in terms of the size of raindrops, their terminal velocity, and the rainfall rate.

11:00

5aSA9. Dynamic analysis of shuttle bus generated vibration in an airport rental car center with an elevated roadway. James Phillips (Wilson, Ihrig & Assoc., 5776 Broadway, Oakland, CA 94618, jphillips@wiai.com)

A finite-element analysis (FEA) model was developed for a new rental car center at an international airport in the United States. Incorporated into the structure was an elevated roadway for shuttle buses transporting rental car customers. Of concern was possible vibration transmitted into occupied portions of the structure from shuttle buses. Measurements were conducted on an existing elevated roadway at the airport in order to determine dynamic force inputs to be applied in the FEA model developed to predict the vibration response inside the rental car center. Impulse response measurements were conducted and then the vibration response of the roadway was measured during shuttle bus passbys. The results of these tests were used to derive the dynamic forces generated by the moving buses on the structure. A finite-element analysis (FEA) model of the elevated roadway was also developed. This model was used to verify the derived dynamic force levels and to verify the modeling approach applied to the rental car center.

11:15

5aSA10. Parametric relations for porous foams and their use in optimization of structural acoustic performance of multilayer trim components. Eleonora Lind and Peter Goransson (KTH Aeronautical and Vehicle Eng., MWL, SE-10044 Stockholm, Sweden, eleonora@kth.se)

Lightweight porous acoustic multilayer trim components have traditionally been specified in terms of sound absorption and sound transmission loss performance targets. Importantly, the material specification of the trim component developed only for absorption and sound transmission loss may be suboptimal in terms of, e.g., sound radiation behavior. This highlights the necessity for accurate, computationally efficient, and robust simulation method, which should form an integral part of a multidisciplinary optimization tool. In addition, for such optimization to be physically meaningful the design parameters used should be based on relations between microstructural dimensions and properties, and the corresponding macroscopic parameters describing porous materials. This paper discusses recent results from research focused on the parametrization of porous foams, i.e., continuous links between micro-dimensions, elasticity, density, and flow resistivity, used in higher order 3-D finite element simulations of multilayer components for structural-acoustic applications. In addition to the microdimensions, also the thicknesses of individual layers have been used as design parameters, with the overall weight as the design objective. Acoustic and vibration targets, velocity of the radiating surface, radiation efficiency, transmission, etc., as well as upper and lower bounds of the microdimensions, have been used as constraints.

5a SAT. AM

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Session 5aSCa

Speech Communication: Speech and Hearing in Targeted Populations (Poster Session)

Mini Shrivastav, Cochair
Univ. of Florida, Dept. of Communications Science and Disorders, 336 Dauer Hall, Gainesville, FL 32611

Hideki Kasuya, Cochair
International Univ. of Health and Welfare, Dept. of Speech, Language and Hearing Science, 2600-1 Kita-kanemaru, Otawara 324-8501, Japan

Contributed Papers

All posters will be on display from 8:00 a.m. to 11:00 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:30 a.m. and contributors of even-numbered papers will be at their posters from 9:30 a.m. to 11:00 a.m.

5aSCa1. Making young ears old and old ears even older: Simulating a loss of synchrony. Ewen MacDonald, Kathy Pichora-Fuller, Bruce Schneider, and Willy Wong (Univ. of Toronto, Toronto, ON Canada M5S 3G9)

Age-related changes in the auditory system have been attributed to three independent factors: OHC damage, changes in endocochlear potentials, and loss of neural synchrony. In previous studies, a jitter algorithm has been used to simulate the loss of synchrony in young adults (MacDonald *et al.*, 2005). In this study, the effect of jitter on old adults with good audiograms in the speech range is explored. SPIN-R sentences were presented in two SNR and three processing conditions: intact, jitter, and smear. The parameters of the jittering algorithm were the same as those used with young adults. The parameters of smearing algorithm were chosen to match the spectral distortion produced by jitter algorithm. While both the jitter and smear conditions resulted in a significant decline in word identification, the decline was largest in the jitter condition. Psychometric functions were fitted to the data and compared to previous work with young adults. The comparison supports the hypothesis that loss of synchrony can adversely affect speech intelligibility in noise, and is consistent with the hypothesis that loss of synchrony occurs with age. As well, the comparison suggests that the effect of jitter may be linear.

5aSCa2. Adult age differences in the use of envelope cues to identify noise-vocoded words with a varying number of frequency bands. Signy Sheldon, Kathy Pichora-Fuller, and Bruce Schneider (Univ. of Toronto at Mississauga, Rm. CCIT 4163, 3559 Mississauga Rd., Mississauga, ON, L5L1C6, Canada, signy.sheldon@utoronto.ca)

Older adults with good audiograms have difficulty understanding speech in noise. Age-related differences have been found on some temporal processing measures such as gap detection; however, older adults are believed to have well-preserved ability to use envelope cues to identify words. Following Shannon *et al.* (1995), we used noise-vocoded speech such that the amplitude envelope of speech was retained in frequency bands but filled with noise, thereby obliterating fine structure cues within each band. In experiment 1, younger and older listeners heard a list of words. Each word was presented first with one vocoded frequency band, and the number of bands was incremented until the listener correctly identified the word. The average number of bands required for correct identification was found to be identical for both age groups. In experiment 2, both age groups identified words in four blocked noise-vocoded conditions (16, 8, 4, and 2 bands). Younger adults outperformed older adults. Although older adults were as able as younger adults to use envelope cues cumulatively in experiment 1, they were less able to use these cues without the benefit of repetition.

5aSCa3. Audiovisual perception of voicing with age in quiet and cafe noise. Dawn Behne (Dept. of Psych., Norwegian Univ. of Sci. and Technol., 7491 Trondheim, Norway), Yue Wang (Simon Fraser Univ., Burnaby, BC V5A 1S6, Canada), Magnus Alm, Ingrid Arntsen, Ragnhild Eg, and Ane Valsø (Norwegian Univ. of Sci. and Technol., 7491 Trondheim, Norway)

Research has shown that voicing is difficult to discern in noisy environments. While voicing may be difficult to resolve from visual cues, acoustic cues for voicing are relatively robust. This study addresses these factors with normally aging audiovisual perception. Identification responses were gathered with 19–30-year-old and 49–60-year-old adults for audiovisual (AV) CVs differing in voicing and consonant place of articulation. Materials were presented in quiet and in cafe noise (SNR=0 dB) as audio-only (A), visual-only (V), congruent AV, and incongruent AV. Results show a tendency toward use of visual information with age and noise for consonant place of articulation. Notably for voicing, incongruent AV materials that had one voiced component, regardless if it was A or V that was voiced, were consistently perceived as voiced in both age groups and regardless of noise. Only if the A and V components were both voiceless was the syllable perceived as voiceless. These findings indicate the influence of age and noise in the use of perceptual information to identify place of articulation. That voicing is robustly salient from either audio or visual information, despite the unlikely presence of strong visual cues for voicing, indicates a possible bias toward the perception of voicing.

5aSCa4. Energy suppression of steady-state portions of vowels while maintaining the energy of consonants better improves speech intelligibility for elderly listeners in reverberation. Yusuke Miyauchi and Takayuki Arai (Dept. of Elec. and Electron. Eng., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554 Japan, m-yuusuk@sophia.ac.jp)

In a reverberant environment, overlap-masking renders speech perception difficult. Arai itet al. proposed that energy suppression of steady-state portions of speech improves speech intelligibility with young subjects [Acoust. Sci. Technol. 23, 229–232 (2002)]. Audibility degrades with age. Therefore, reverberation can be a more critical barrier to speech perception by elderly listeners. To investigate the effect of suppressing reverberation for elderly people, a listening test was conducted with 25 elderly

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subjects (mean age: 73.3 years) using the following three types of speech: (a) speech without energy suppression of steady-state portion, (b) speech with the suppression of steady-state portions of speech, and (c) speech with the suppression only of vowels, while maintaining the energy of consonants. As expected, speech intelligibility by method (b) (46.7%) was markedly improved from (a) (42.2%), but method (c) (52.0%) improved it more. Because the energy of consonants is less than that of vowels, overlap-masking attendant on previous vowels would largely affect perception of subsequent consonants. These results suggest that suppressing the steady-state portions of vowels while maintaining the energy of consonants serves elderly persons well to improve speech intelligibility in a reverberant environment. [Work supported by JSPS.KAKENHI (16203041).]

5aSCa5. Older adult's identification and memory of synthetic and natural speech in noise. Candice Q. McCarty and Aimee Surprenant (Purdue Univ., 703 Third St., West Lafayette, IN 47907)

This research tested older adult's performance on identification and recall when presented sentences taken from the Speech Perception in Noise Test [Bilger *et al.*, J. Speech Hear. Res. **27**, 32–48 (1984)] synthesized using AT&T Natural Voices speech synthesizer. Performance on synthesized speech was compared to identification and recall of natural speech in noise. In experiment 1, participants were instructed to identify the final word of high- and low- predictability natural and synthesized sentences presented at three signal-to-noise ratios. The results showed that natural speech was easier to identify than synthesized speech at every speech-to-noise ratio for high and low predictability sentences. In addition, replicating Pichora-Fuller *et al.* [J. Acoust. Soc. Am. **97**, 593–608 (1995)], pure tone thresholds were significantly correlated with identification performance for both natural and synthetic speech. In experiment 2, participants were given a sentence span task in which they were presented two to five natural and synthesized sentences and asked to recall the final words. These experiments show that, even with high context, synthetic speech is more difficult to understand and remember than natural speech. These results highlight concerns about the use of synthesized speech in assistive devices, particularly for elderly listeners. [Work supported by NIH.]

5aSCa6. Spectral shape discrimination and speech identification in noise in elderly hearing-impaired listeners. Mini Shrivastav (Dept. of Commun. Sci. and Disord., Univ. of Florida, 336 Dauer Hall, Gainesville, FL 32611)

Elderly hearing-impaired listeners have difficulties in the recognition and discrimination of consonants, particularly those that share the same manner of articulation. An important cue that distinguishes the manner of articulation of these consonants is spectral shape. A previous study (Shrivastav *et al.*, 2006) found a moderate predictive relation between spectral-shape discrimination thresholds and syllable identification scores of elderly hearing-impaired listeners, when all stimuli were presented in quiet. The present study examined the contribution of spectral-shape discrimination abilities to speech-identification performance of elderly hearing-impaired listeners in the presence of background noise. The study included a group of elderly hearing-impaired listeners, with a group of young normal-hearing adults included for comparison purposes. Listeners were tested on a series of speech-identification and spectral-shape discrimination tasks, while ensuring that all stimuli were at least minimally audible to all the hearing-impaired listeners. The contribution of spectral-shape discrimination abilities to speech-identification performance in background noise was examined and compared to the results in quiet for young normal-hearing and older hearing-impaired listeners.

5aSCa7. The use of subsegmental information in sentence comprehension with or without formant transitions by normal-hearing and hearing-impaired listeners. Jae Hee Lee and Diane Kewley-Port (Dept. of Speech and Hearing Sci., 200 S. Jordan Ave., Bloomington, IN, 47405, jaejalee@indiana.edu)

Using a noise replacement paradigm, Lee and Kewley-Port [J. Acoust. Soc. Am., **119**, (2006)] examined sentence comprehension by young normal-hearing (YNH) and elderly hearing-impaired (EHI) listeners when sentences were processed to present only subsegmental information (i.e., either steady-state or formant transitions) in sentences processed in four different ways. Results showed that correct word responses by EHI listeners were more affected by the type of information in the four conditions compared to YNH listeners. To compare the use of subsegmental information between listener groups who were carefully matched for audibility and age, new analyses of correctly identified phonemes and different word-error patterns were made for both correct and incorrect word responses. Correlation analyses examined the relations among correct and incorrect word responses, phoneme scoring, hearing thresholds, and age. Despite a larger number of incorrect answers by EHI listeners, word-error patterns, as well as an ability to extract phonemes using subsegmental information, were similar between YNH and EHI listeners. Analyses for EHI listeners showed correct word responses were strongly correlated with correct phoneme identification and with hearing thresholds, but not with age. [Work supported by NIHDCD-02229.]

5aSCa8. Auditory-visual integration and lipreading abilities of older adults with normal and impaired hearing. Mitchell S. Sommers (Dept. of Psych., Washington Univ., Campus Box 1125, St. Louis, MO 63130, msommers@wustl.edu), Nancy Tye-Murray, and Brent Spehar (Washington Univ. School of Medicine, St. Louis, MO 63130)

The current study investigated how age-related hearing impairment affects lipreading and auditory-visual integration in older (above age 65) individuals. The performance of 53 normal-hearing and 24 mild-to-moderate hearing-impaired older adults was compared on auditory-only (A), visual-only (V), and auditory-visual (AV) speech perception, using consonants, words, and sentences as stimuli. All testing was conducted in the presence of individually specified multi-talker background babble, to obtain approximately equivalent A-only performance across the groups. In addition, we compared normal-hearing and hearing-impaired individuals on measures of auditory enhancement, visual enhancement, and auditory visual integration that were derived from the A, V, and AV performance scores. In general, normal-hearing and hearing-impaired older adults performed similarly on measures of visual-only and auditory-visual speech perception. The one exception to this finding was that hearing-impaired adults performed significantly better than normal-hearing participants on visual-only identification of words. Measures of visual enhancement, auditory enhancement, and auditory-visual integration did not differ as a function of hearing status. Overall, the results of the current study suggest that despite increased reliance on visual speech information, hearing-impaired older adults do not exhibit better visual-only speech perception or auditory-visual integration than age-matched normal-hearing individuals.

5aSCa9. Effects of intermodal timing difference and speed difference on auditory-visual speech perception. Akihiro Tanaka (Dept. of Psych., Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, 113-0033 Tokyo, Japan, tanaka@L.u-tokyo.ac.jp), Shuichi Sakamoto, Komi Tsumura, and Yo-iti Suzuki (Tohoku Univ., Sendai 980-8577, Japan)

Previous studies have shown that lipreaders possess insufficient sensitivity to intermodal timing differences and adapt to the timing difference when it is kept constant. This study investigated effects of intermodal timing differences and speed differences on auditory-visual speech perception. We used 20 minimal pairs of Japanese four-mora words such as *mizuage* (catch landing) versus *mizuame* (starch syrup). We administered intelligibility tests to younger and older adults. Words were presented

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under visual-only, auditory-only, and auditory-visual (AV) conditions. Effects of AV asynchrony by time-expanded speech (speed difference) on AV benefit (i.e., the superiority of AV performance over auditory-only performance) were compared with those by intermodal timing shift (timing difference). Results showed that the AV benefit declined as the timing difference and speed difference increased. Results also revealed that younger adults adapted to the timing difference but did not adapt to the speed difference. Older adults adapted neither to the timing difference nor to the speed difference. These results suggest that intermodal lag adaptation in younger adults requires a constant timing difference between a talker's moving image and voice. These results might be useful for design of a multimodal speech-rate conversion system.

5aSCa10. Description of mouth shapes used in traditional Japanese sign language by comparison to those of speech utterance. Yumiko Fukuda (Natl. Rehabilitation Ctr. for the Disabled, Tokorozawa, Japan, fukuda@rehab.go.jp) and Shizuo Hiki (Waseda Univ., Tokorozawa, Japan)

Mouth shapes used in traditional Japanese sign language were described systematically through comparison to those in uttered speech. A framework of mouth shape descriptions that had been proposed by the present authors included (1) projection of mouth shapes of Japanese vowels and semi-vowels to the lip articulation plane of the three-dimensional IPA vowel chart and (2) grouping of discriminating consonants in lip-reading on the IPA articulation matrix. It was transferred to a coordinate consisting of contractions of muscles involved in mouth movement. In traditional Japanese sign language, the mouth shape, when added as a part of facial expression, modifies meaning of the word conveyed by a hand gesture. Those mouth shapes were extracted from conversations among native signers and were symbolized by their respective degrees of closing, rounding, and pulling corners, and by categories borrowed from Japanese vowels and some consonants. Using the extended framework, those mouth shapes that were elected for visual discrimination ranged more widely than those elected to discriminate phonemes aurally. Video pictures were retrieved according to their symbols from an electronic dictionary of 250 basic sign words and 1500 exemplary sentences of their meanings, as edited by the Japanese sign language research group, NRCD.

5aSCa11. Audiovisual speech integration in children with autism spectrum disorders. Julia R. Irwin (Haskins Labs., 300 George St., New Haven, CT 06511)

Children with autism spectrum disorder (ASD) appear to be less influenced by visual speech information than typically developing children, as measured by their responses to mismatching auditory and visual (McGurk) stimuli. This study examined whether this reduction in sensitivity to the McGurk effect is due to eye gaze aversion, a hallmark of ASD. Children with ASD and typically developing controls (TD) were presented with videotaped consonant-vowel (CV) stimuli. Stimuli were digitally edited to create either an audiovisual (AV) match (AV /ma/ or /na/) or mismatch (audio /ma/ and visual /ga/). Responses were considered visually influenced if participants reported hearing /na/ for the mismatched stimuli. Using eye-tracking methodology, only trials where participant's gaze was fixated on the speaker's face during consonantal closure were included in analyses. Initial analyses reveal, when fixated on the speaker's face, children with ASD show significantly less visual influence relative to typically developing controls. It is proposed that the reduced influence of visual speech information in ASD is best understood at the level of visual speech processing, perhaps arising from a lack of experience with speaking faces. Implications for understanding visual speech processing in typical development and developmental disability are discussed. [Work supported by NIH grant DC-00403 (Cathi Best, PI.]

5aSCa12. Auditory temporal order judgments in children with specific language impairment and autism spectrum disorder. Sheryl Rosen, Mark De Ruiter, and Li Hsieh (Wayne State Univ., 207 Rackham, 60 Farnsworth, Detroit, MI 48202)

The Tallal repetition test (TRT) measures the ability of listeners to detect, associate, sequence, and remember complex patterns of stimuli (serial memory) The TRT contains sequencing and serial memory subtests with varying short and long interstimulus intervals (ISIs) Three groups of 12 children were tested using a modified version of the TRT. Subject groups consisted of 7- to 10-year old children with typical development (TD), autism spectrum disorder (ASD), and specific language impairment (SLI). The TRT contained both tone and CV syllable stimuli to assess temporal processing. As ISI increased, performance improved for both children with TD and SLI in both tone and syllable sequencing tasks, but not for children with ASD, especially in the tone task. Children with TD and those with ASD performed better than those with SLI across most subtests, especially in tasks with short ISIs. Analysis of performance in the serial memory portion of the TRT revealed a predicted pattern of decreased performance as stimuli elements increased. Overall, children with TD and those with ASD appear to have superior auditory temporal order judgment abilities when compared to children with SLI. The linguistic relevance factor shall be considered in assessment and training of various disordered populations.

5aSCa13. Perception of synthesized syllables and sentences by listeners with hearing loss. Corine Bickley, Amanda Dove, Andrea Liacouras, and Jolene Mancini (Dept. of Hearing Speech and Lang. Sci., Gallaudet Univ., 800 Florida Ave NE, Washington DC 20002)

The goal of this project is to create a set of synthesized speech materials that yields listening results equivalent to those that would be obtained using recorded naturally spoken materials, such as the CUNY Nonsense Syllable Test and the CASPERsent set of topic-specific sentences. One research interest of the RERC-HE is the following: Can synthesized speech be used as a substitute for recorded human speech in aural rehabilitation and in hearing aid research? The listeners in this study are both adults with normal hearing and with hearing loss. We have tested two listeners with normal hearing and four with hearing loss using nonsense syllables synthesized with DECtalk. The patterns of errors in their identification of phonemes follow expected patterns when subjects' responses were analyzed with respect to their audiometric profiles and the acoustic characteristics of the synthesized phonemes. The same group of listeners is now being tested using sentences from the CASPERsent lists synthesized with two commercially available text-to-speech systems: DECtalk, which uses a formant-type synthesizer, and ATT Natural Voices, which uses a concatenative synthesizer. Results of these listening tests will be presented. [This project is funded through the Rehab Engineering Research Center—Hearing Enhancement through the NIDRD.]

5aSCa14. Effects of dichotic listening for hearing impaired people. Ai Sasaki, Atsunobu Murase, Shuichi Sakamoto, Yôiti Suzuki (Res. Inst. of Electron. Commun., Grad. School of Info. Sci., Tohoku Univ. Katahira 2-1-1, Aoba-ku, Sendai, 980-8577 Japan, asasaki@ais.riec.tohoku.ac.jp), Tetsuaki Kawase, and Toshimitsu Kobayashi (Tohoku Univ., Aoba-ku, Sendai, 980-8577 Japan)

Methods applying the dichotic listening algorithm have been proposed and examined for application in hearing aids. Murase *et al.* (2005) proposed an algorithm in which dichotic listening was simply implemented by a set of complementary high-pass and low-pass filters. One filter is for either the right or left channel; the other filter is used for the other channel. This study examined two unsolved problems with this algorithm. One is the reason why this algorithm is effective. The other is probable degradation of sound localization that occurs because interaural level differences and interaural time differences is only slightly available with this signal processing. This problem is considerable when the sounds come from frontal incidence. Regarding the first problem, intelligibility tests were

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performed using the low-frequency-boosted sounds. The intelligibility scores under the dichotic listening conditions should be higher than those under the diotic listening conditions if the effectiveness of this algorithm is attributable to the reduction of upward spread of masking. Results suggest that this dichotic listening reduced the upward spread of masking. Regarding degradation of performance of sound localization, sound localization test results suggest that time-intensity trading might recover degradation with dichotic listening algorithm.

5aSCa15. Effects of auditory training on hearing aid acclimatization. Jack M. ScottIII (Texas Medical & Surgical Assoc., 8440 Walnut Hill Ln., Ste. 500, Dallas, TX 75231, jack.scott@texasmedsurg.com)

New hearing-aid users may experience a gradual improvement in speech performance over time that typically plateaus 6–12 weeks following hearing aid fitting. The delayed improvement may be due to neural reorganization of the auditory system following the introduction of amplified auditory input. Unfortunately, 30 days is the typical duration for most hearing aid trial periods; therefore, many individuals may be returning their hearing aids before being able to evaluate the full benefit of amplification. Prior to amplification, listeners with hearing impairment may experience difficulty perceiving subtle high-frequency changes in speech known as second-formant transitions. Second-formant transitions provide cues in discriminating consonant sounds. The purpose of this study was to determine if intensive high-frequency discrimination training for new hearing-aid users could facilitate improvements in speech perception. Speech recognition was evaluated for consonant identification, speech feature perception, and sentence identification in quiet and in noise before and after 10 days of training discrimination of frequency sweeps at 2 kHz. Improvements were found in sentence identification. These results have implications for the use of auditory training to improve speech recognition for newly amplified listeners with hearing impairment.

5aSCa16. Vowel production in Swedish children with cochlear implants. Tina Ibertsson, Birgitta Sahlen, and Anders Lofqvist (Dept. Logoped., Phoniatrics & Audiol., Clinical Sci., Lund, Univ. Hospital, SE-221 85 Lund, Sweden)

This study analyzed vowel production of Swedish children with cochlear implants. Swedish has a rich vowel system with nine vowels. Minimal word pairs were used as linguistic material. The age of the children varied between 6 and 13 years. The age at implantation was 1.2–10 years. The first two formants were measured for each vowel. To estimate the vowel space, the average Euclidean distance in the $F1$–$F2$ plane between each vowel and the mean first and second formant frequencies of all the vowels was calculated for each child. Preliminary results from 15 children with cochlear implants and 11 control subjects with normal hearing show no difference between the two groups, due to a large amount of variability in the group with cochlear implants. The relationship between the vowel space and the speech perception of each child with a cochlear implant was analyzed using correlations. Results show no strong correlation between the two measures, again due to the variability within the group. [Work supported by the Sunnderdahl Foundation.]

5aSCa17. Sentence duration and mode of communication in cochlear implanted children. Nicole L. Wiessner, Emily A. Tobey (Callier Adv. Hearing Res. Ctr., Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235), and Ann E. Geers (Univ. of Texas Southwestern Medical Ctr. at Dallas, Dallas, TX 75390)

This study investigates the relationships between mode of communication and speech duration in cochlear implanted (CI) children with 4–6 years experience with their device. One hundred thirty-six CI children between the ages of 8 and 9 years old repeated sentences of three, five, and seven syllables (McGarr, 1983). Durations were examined according to syllable length, low versus high context sentences, and the communication modality of the child's classroom. Findings suggest that communication modality is more strongly associated with duration than sentence context or length. Significantly shorter sentence durations were found for CI children taught with an auditory-oral emphasis than CI children taught using an emphasis of speech and some form of manual sign system. [Work supported by the NIH.]

5aSCa18. Interpolation of vocal-tract shape during stop closures from transition segments in vowel-consonant-vowel syllables. Milind S. Shah and Prem C. Pandey (Dept. of Elec. Eng., Indian Inst. of Technol., Bombay, Powai Mumbai 400076, India, pcpandey@ee.iitb.ac.in)

Children with prelingual profound hearing impairment have great difficulty in acquiring speech. Speech-training systems providing visual feedback of vocal-tract shape are found to be useful for improving vowel articulation. Vocal-tract shape estimation, based on LPC and other analysis techniques, generally fails during stop closures, and this restricts its effectiveness in speech training for production of consonants not having visible articulatory efforts. A technique based on two-dimensional surface modeling of the area values, estimated by LPC analysis, during the vowel-consonant and consonant-vowel transitions preceding and following the stop closure, has been investigated for interpolating the area values during the stop closures. Surface modeling was based on least-squares bivariate polynomials and Delaunay triangulation methods. Syllables of the type /aCa/, /aCi/, /iCa/, /iCi/, and /uCu/ with stop consonants /p/, /b/, /t/, /d/, /k/, and /g/ were analyzed for the estimation of place of stop closure. For bilabial, alveolar, and velar stops, the place could be estimated consistently with conic polynomial surface interpolation. Estimation of place, based on Delaunay surfaces, was consistent for bilabial and velar stops. However, cubic polynomial surface interpolation results were less consistent in estimating the place of constriction.

5aSCa19. Child voice and noise; acoustic effects of a day at the day-care related to background noise levels. Anita McAllister (INR, Speech Pathol., Linkoping Univ., Sweden), Svante Granqvist, Johan Sundberg, Peta Sjolander (Royal Inst of Technol.), and Mechtild Tronnier (Linkoping Univ.)

Several studies have found that high background noise levels are detrimental to health parameters. In particular, this seems to apply to developing voices were future vocal habits are established. Thus, it is important to study vocal function and environmental effects on the developing child voice. This study analyzed the effects of background noise on children's voices, specifically vocal intensity, and fundamental frequency. The investigated vocal parameters were (1) the relationship of background noise levels to F0 and vocal intensity, (2) F0 and vocal intensity variations over the day, and (3) F0 perturbation variations over the day. Ten 5-year-old children from three day-cares participated, six boys and four girls. The audio signal was recorded by two microphones mounted in front of the subjects' ears. By adding these signals it is possible to separate the voice from background noise. The material analyzed contained data from three 60-min recordings per child from morning, noon, and afternoon during a normal day at the day-care. Generally high mean background noise levels were found (82.6 dBA). Preliminary results suggest a correlation between high background noise and high F0 and vocal intensity in the children's voices, particularly for boys. F0 perturbation tends to increase during the day.

5aSCa20. Acoustic correlates of primary motor speech disorders in children during oral and hand tasks. Beate Peter and Carol Stoel-Gammon (Univ. of Washington, 1417 N.E. 42nd St., Seattle, WA 98195)

Primary child speech disorders have been subclassified variously. One proposed subtype, childhood apraxia of speech (CAS), results from motor programming deficits in speech production, and associated characteristics include limited phoneme inventory, variability in production, and diffi-

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culty with speech prosody, e.g., monostress and monopitch. The purpose of this study was to identify acoustic correlates of oral and limb movements, derived from the hypothesis of a generalized timing deficit, that characterize this disorder subtype differentially from other disorder subtypes. Twelve children (age 4;7 to 6;6), of which eight were labeled with CAS, and 12 age- and gender-matched peers participated in a variety of oral motor and hand tasks. *K*-means clustering generated three disorder subgroups in the sample of participants with speech disorders, two of which were populated with CAS labels. Distinguishing characteristics included accuracy of vowel duration in a nonword imitation task, accuracy of clap interval duration in a clapped rhythm imitation task, rate of syllable repetition, durations of unstressed but not stressed vowel durations, and breath group duration. Taken together, these results are consistent with the view that deficits in timing accuracy, movement speed, and breath support underlie at least one subtype of idiopathic speech disorder.

5aSCa21. Changes in acoustic characteristics of /r/ following production training using electropatatography. Anna Schmidt (School of Speech Pathol. & Audiol., Kent State Univ., Kent, OH 44242)

Visual feedback using electropalatography (EPG) is an effective method for changing production of speech sounds in children. Differences in tongue placement and general acceptability of production have been demonstrated with EPG training for tongue to palate contact. Little is known, however, about changes in the spectral characteristics of speech sounds following training. The current study examined pre- and post-EPG patterns for /r/ in prevocalic, intervocalic, and postvocalic positions in sentences with 5 weeks of intensive EPG training. In particular, pre- and post-spectral characteristics of /r/ were examined for differences pre- and post-treatment. Subjects were four children (aged 9–10). These children exhibited distorted /r/ despite several years of traditional speech therapy but learned to produce perceptually acceptable /r/ with EPG. Spectral characteristics for /r/ productions in the same sentences by four typically developing children were used for comparison. Differences will be discussed in terms of acoustic characteristics as well and changes in tongue position.

5aSCa22. Physiologic and behavioral classification of delayed speech: Syllable repetition tasks. Jennell C. Vick, Christopher A. Moore, Lakshmi Venkatesh (Dept. of Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, jennell@washington.edu), Thomas F. Campbell, Heather Leavy Rusiewicz (Univ. of Pittsburgh, Pittsburgh, PA), Lawrence D. Shriberg (Univ. of Wisconsin—Madison, Madison, WI), and Jordan R. Green (Univ. of Nebraska—Lincoln, Lincoln, NE)

As part of a large-scale study that aims to create a data-driven model of speech delay of unknown origin (SD) in preschool children, this investigation analyzed two- and three-syllable nonword repetitions by 50 3- to 5-year-old children, half of whom are typically developing (TD). The syllable repetition task was composed of four different words, bama, bada, bamana, and manaba, selected both for articulatory simplicity and to maximize displacement of the external articulators for kinematic analyses. While children in the TD group were significantly more accurate in both two- and three-syllable repetitions than children in the SD group, neither group was able to repeat three-syllable nonwords with particular acumen. Children in the TD group were 55% accurate, while children in the SD group were only 22% accurate. Measures of repetition accuracy, acoustics, nasalence, external articulatory kinematics, and respiration contribute to a hierarchical agglomerative clustering model that generates hypotheses about potential subclinical categories within the SD and TD groups.

5aSCa23. Changes of voice characteristics under noisy conditions: Acoustic and fiberscopic studies of school teachers. Noriko Kobayashi (School of Allied Health Sci., Kitasato Univ., 1-15-1, Kitasato, Sagami-hara, Kanagawa, 228-8555 Japan, noriko@ahs.kitasato-u.ac.jp) and Takashi Masaki (Kitasato Univ, Sagamihara, 228-8555 Japan)

It has been known that incidence of vocal pathology was high in school teachers as they had to use a great amount of voice and loud phonation in noisy school settings with the children's loud voices. Vocal hygiene programs and voice therapy are effective treatments for these cases. The final goal of the treatment should be to obtain the skills to use efficient and healthy phonation even in abusive and noisy environments. In our study, two school teachers with vocal nodules, two speech therapists who were trained to keep proper phonation even in noisy environments, and ten college students with no laryngeal pathology (control group) spoke eight sentences in two environments: a quiet environment and one with meaningful multitalker babble (MMB). Acoustic analyses and fiberscopic examination revealed higher sound pressure and F0 levels and laryngeal constriction in MMB for the teachers before therapy and the students, whereas two speech therapists kept similar F0 levels and the nonconstricted laryngeal condition in both environments. A teacher who had voice therapy produced less constricted voice with MMB than the pretherapy recording. The results suggested the efficacy of voice therapy with MMB for teachers to obtain efficient and healthy voice in their noisy work environments.

5aSCa24. Effect of speech task on intelligibility in speakers with dysarthria. Kate Bunton (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721-0071, bunton@u.arizona.edu)

There is some evidence that speech intelligibility varies across speech task for speakers with dysarthria. Primary differences have been related to the linguistic level of the material being evaluated (e.g., single words, sentences, conversation). While conversational speech is the most socially valid context for evaluating speech intelligibility, quantifying intelligibility in conversation limits its clinical use. The present study assessed intelligibility in speakers with dysarthria across four speech production tasks: single words, sentence production, passage reading, and spontaneous speech. Speakers varied with regard to the type of dysarthria, including hypokinetic, hyperkinetic, spastic, and mixed. Results show that all speakers were significantly less intelligible during spontaneous speech than the other structured speech production tasks (e.g., single words, sentences, and reading). Mean differences varied across speech severity level but did not vary across dysarthria type. These data indicate that speech production task is an important variable to consider during the evaluation of dysarthria and highlight the importance of developing techniques to measure the intelligibility of conversational speech. [Work supported by NIH Grant R03 DC005902.]

5aSCa25. Acoustic features of speech after glossectomy. Yuichi Ishimoto (School of Media Sci., Tokyo Univ. of Technol., 1404-1 Katakura, Hachioji, Tokyo 192-0982, Japan, ishi@media.teu.ac.jp) and Noriko Suzuki (Showa Univ., Ohtaku, Tokyo 145-8515, Japan)

The tongue serves an important role in speech production. Speech disorders due to tongue resection often occur and cause communication difficulties after surgery, according to the extent of the resected area. Such patients typically require appropriate speech therapy and palatal augmentation prosthesis. However, diagnosis of the speech in such patients has been based largely on subjective evaluation by clinicians. Acoustic features in glossectomee have yet to be clarified, and there is no objective estimation method for speech intelligibility. This study aims to clarify acoustic features associated with speech disorders after glossectomy and to develop a computerized evaluation system. The speech data were analyzed using the improved cepstrum method and compared with varying degrees of speech intelligibility. The vowel /i/ was selected as the speech sample because it is the high front vowel. The results showed that spectral

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envelopes of the Japanese vowel /i/ with low speech intelligibility exhibit two peaks in the second formant (F2) region from 1500 to 3000 Hz. This suggests that the spectral features in the F2 region of the vowel /i/ are important clues to determine speech intelligibility.

5aSCa26. New generation aids for laryngectomy patients. Andrzej Czyzewski, Piotr Odya (Gdansk Univ. of Technol., Narutowicza 11/12, PL-80-952 Gdansk, Poland), and Bozena Kostek (Inst. of Physiol. and Pathol. of Hearing, Warsaw, Poland)

The artificial larynx has many disadvantages. The produced speech is monotonous and sounds artificially. In addition, produced speech intelligibility is usually poor. The major problem is a background noise caused by the device. In fact, the artificial larynx is only a simple vibrator, a construction of which has been almost unchanged since the 1950s. The aim of the presented project is to design a new generation of devices for laryngectomy patients. There are two different approaches to solve this task. The first one focuses on the artificial larynx. Some major improvements in the construction of the device might be easily introduced. Hence, the artificial larynx engineered was equipped with a digital processor and an amplifier. The spectral subtraction algorithm for noise reduction was used. In this method, an average signal spectrum and average noise spectrum are estimated and subtracted from each other, thus average signal-to-noise ratio (SNR) is improved. The main problem to be solved was that both the noise and the speech signal have the same excitation source and consequently are strongly correlated for voiced sounds. The second approach uses a PDA (portable digital assistant) to generate synthetic speech. The proposed new generation devices helping laryngectomy patients are presented in the paper.

5aSCa27. A speech communication aid system for total laryngectomies using voice conversion of body transmitted artificial speech. Keigo Nakamura, Tomoki Toda, Hiroshi Saruwatari, and Kiyohiro Shikano (Nara Inst. of Sci. and Technol., 8916-5, Takayama-cho, Ikoma-shi, Nara, 630-0192 Japan, kei-naka@is.naist.jp)

There are several problems associated with using existing electrolarynxes. For example, the loud volume of the device itself might disturb smooth interpersonal communication, and its generated speech is also unnatural. To improve the quality of speech communication using such a medical device, this paper proposes a novel speech communication aid system for total laryngectomies. This system detects articulated speech caused by a new sound source as an alternative to the existing electrolarynx through the soft tissues of the head with a nonaudible murmur (NAM) microphone attached to the surface of the skin [Nakajima *et al*, Proc. Interspeech 2005, pp. 293–296 (2005)]. The new sound source outputs signals of extremely low energy that cannot be heard by people near the speaker. Such body-transmitted artificial speech is converted to a more natural voice by statistical voice conversion [T. Toda and K. Shikano, Proc. Interspeech 2005, pp. 1957–1960 (2005)]. The performance of the proposed system is evaluated in terms of objective and subjective measures using body-transmitted artificial speech simulated by a non-disabled speaker. Experimental results show that body-transmitted artificial speech is consistently converted to a much more natural and intelligible voice. [Work supported by SCOPE-S.]

5aSCa28. Development of a hands-free electrolarynx for persons whose larynx have been removed. Mitsuo Hashiba (Hokkaido Industrial Res. Inst., N19W11, Kita-ku, Sapporo, 060-0819, Japan), Yasunori Sugai (Densei Inc., Chuo-ku, Sapporo, 060-0002, Japan), Takashi Izumi (Hokkaido Tokai Univ., Minami-ku, Sapporo, 005-8601, Japan), Shuichi Ino, and Tohru Ifukube (The Univ. of Tokyo, Meguro-ku, Tokyo, 153-8904, Japan)

An electrolarynx is a useful speech-substitute device for patients who have lost laryngeal function. Unfortunately, the voices produced by a conventional electrolarynx are completely flat, unlike a human voice. For that reason, we developed a pitch-controlled electrolarynx that allows patients to control a voice intonation using their exhalation [N. Uemi and T. Ifukube, "Design of a new electrolarynx having a pitch control function," IEEE RO-MAN'94, 198–203 (1994)]. This pitch-controlled electrolarynx has been manufactured and more than 3000 patients have used the device since 1998. However, many patients have strongly requested that we design a hands-free electrolarynx so that they can perform daily life functions and work in their office without their hands occupied by the electrolarynx control during use. In response to that request, we designed some models of a hands-free electrolarynx that is attachable to the patient's neck. Furthermore, by comparing the models based on evaluation tests regarding its sound quality and usability, a thermo-plastic brace was found to be the best for attachment to the neck. Although our hands-free electrolarynx has not yet been equipped with the pitch-control function, it was ascertained that it is useful for taking notes while talking on the phone, and so on.

5aSCa29. Modeling aspects of vocal fold oscillations with validation to clinical data. Sarah A. Bentil (Dept. of Mech. Eng., Univ. of Hawaii Manoa, Honolulu, HI 96822), Todd Reed, John S. AllenIII (Univ. of Hawaii Manoa, Honolulu, HI 96822), and Yuling Yan (Stanford Univ. School of Medicine, Stanford, CA 94305)

We investigate vocal fold vibrations in patients with muscle tension dysphonia (MTD) and Parkinson's diseases (PD) using a refined two mass model [I. Steinecke and H. Herzel, J. Acoust. Soc. Am. **97**(3), 1874–1884 (1995)] and clinical data obtained from acoustic recordings and high-speed digital imaging (HSDI) of the larynx. The two-mass model with empirical parameters is able to reproduce some of the behaviors revealed from functional, image-based analyses of vocal fold dynamics in both normal and abnormal states. In particular, the glottal area waveform (GAW) is used to describe the vocal fold vibratory behavior, which is derived from HSDI data using an analytical framework developed by the Yan research group [Yan *et al.*, J. Voice **19**(2), 161–175 (2005); IEEE Trans. Biomed. Eng., (2006)]. Further, a comparison of the two-mass model predictions and the GAW and acoustic data was performed using signal processing techniques including fast fourier transforms, wavelet transforms, and spectrograms. These analyses deliver a comprehensive quantitative representation of the dynamic characteristics of the vocal folds and may provide new insights into the mechanism of abnormal voice productions such as in MTD and PD. [Research is sponsored by NSF Grant #0627992.]

5aSCa30. Preshearing effects on viscoelastic properties of collagen-based vocal fold injectables. Sarah A. Klemuk (Dept. of Speech Pathol. & Audiol., Univ. of Iowa, 336C SHC, Iowa City, IA 52242, sarah-klemuk@uiowa.edu) and Ingo R. Titze (Univ. of Iowa, Iowa City, IA 52242)

A criterion for developing a biomaterial is that its viscoelastic properties correspond to native tissues. This criterion is particularly important for vocal fold tissues. Collagen-based injectables are routinely used to repair paralyzed or scarred vocal folds and restore vocalization, yet reported viscoelastic properties are inconsistent. The present study evaluated the effects of preshearing samples on their viscoelastic measurements. Non-cross-linked bovine collagen (Zyderm II^{TM}) and micronized dermal tissue (Cymetra^{TM}) were presheared by extruding directly through a mechanically operated syringe with different needle sizes. Preshear rates ranged from 31 to 5600/s for orifice radii of 1.21 to 0.0955 mm, respectively. Sandpaper-coated, parallel plate attachments on a stress-controlled rheometer were used to measure elastic moduli and dynamic viscosities from 0.01 to 10 Hz. A 3–10-fold reduction in elastic modulus and a 2–4-fold reduction in dynamic viscosity occurred for preshear rates exceeding 1000/s compared to rates of 31 and 131/s. These results suggest a change in the molecular structure and the viscoelastic compatibility of vocal fold injectables when sufficiently presheared. [Work supported by NIH.]

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5aSCa31. Spasmodic dysphonia in Japan: 1486 Botulinum toxin injections for 260 spasmodic dysphonia patients. Masanobu Kumada (Kumada Ear-Nose-and-Throat Clinic, L-First Bldg. 3F, 4-2-6 Nishi-Azabu, Minato-Ku, Tokyo, 106-0031 Japan) and Takeo Kobayashi (Teikyo Univ. Ichihara Hospital)

Botulinum toxin injection is a very effective treatment for spasmodic dysphonia. We experienced 1486 injections for 260 patients in past 15 years in Japan. Here we show statistics of these injections. We use type A toxin that is produced by Clostridium Botulinum. This toxin affects the neuro-muscular junction as a blocker of release of Ach, which leads to inversible parasis of a muscle. Mainly, we inject the toxin paracutaneously on the neck to the vocalis muscle using a partly covered needle that works as an EMG electrode. Mainly we have four options: 2.5 i.u. unilaterally (38% of 256 patients), 2.5 i.u. for each laterality bilaterally (16%), 5.0 i.u. unilaterally (29%), and 5.0 i.u. for each laterality bilaterally (5%). Its effectiveness persisits for 17.3 weeks on average. The longest effectiveness we have ever experienced is more than 1 year. A small percentage of the injections are not effective; it can be thought that the injected toxin does not reach the neuromuscular junction even when the needle is in the muscle: neuromuscular junctions contribute a part of the muscle and EMG signals do not tell us where these junctions are. Mainly, the side effects are breathy hoarseness and misdiglutition, which persist a week or so.

5aSCa32. Collagen injection as an adjunct to arytenoid adduction. Miwako Kimura, Takaharu Nito (Dept. of Otorhinolaryngology, The Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo, 1138655, Japan, mkimu-tky@umin.ac.jp), Niro Tayama (Intl. Medical Ctr., Shinjuku-ku, 162-8655, Tokyo, Japan), and Roger Chan (Texas Univ., TX)

Dysphonia associated with laryngeal paralysis may be identified in the short term postoperatively or may develop years after successful arytenoid adduction. In selected cases, collagen injection of vocal fold improves phonation after arytenoid adduction. Our experience with the use of collagen injection to supplement arytenoid adduction is reported. Forty patients were treated with transoral collagen injection into the atrophic vocal fold after arytenoid adduction at The University of Tokyo from January 1990 to December 2005. These patients all had a diagnosis of unilateral vocal fold paralysis. Voice quality (GRBAS scale) and the other examinations for vocal function demonstrated measurable improvement after treatment. After arytenoid adduction, maximum phonation time (MPT) improved from 5 to 11 s and mean flow rate (MFR) as an aerodynamic examination improved from 772 to 322 ml/s. Voice quality, especially roughness, improved from 1.8 to 0.8 after collagen injection. MPT (from 11 to 14 s) and MFR (from 322 to 233 ml/s) showed overall improvement after collagen injection. We will report the ease with which transoral collagen injection was accomplished and how better voice quality was attained without the more extensive surgery. In the session techniques of anesthesia, injection, and patient selection are discussed.

5aSCa33. Numerical simulation of sound originated from the vocal tract in soft neck tissues. Makoto Otani, Tatsuya Hirahara (Dept. of Intelligent Systems Design Eng., Toyama PRef. Univ., Imizu, Toyama 939-0398, Japan, otani@pu-toyama.ac.jp), and Seiji Adachi (Fraunhofer Inst. for Bldg. Phys., 70569 Stuttgart, Germany)

A nonaudible murmur (NAM), a very weak speech sound produced without vocal vibration, can be detected by a special NAM microphone attached to the neck, thereby providing a new communication tool for use with functional speech disorders. The microphone is a condenser microphone covered with soft-silicone impression material that provides good impedance matching with soft tissues that are found in the neck. The NAM detected with the NAM microphone, however, is insufficiently clear because higher-frequency components are suppressed severely. A production mechanism of the NAM as well as transfer characteristics of the NAM in soft tissues of a neck should be clarified to improve the NAM clarity. As a first step, sound propagation from the vocal tract to the neck surface is simulated using a head model scanned with magnetic resonance imaging and finite difference method. Results show that soft-meat-tissue-conducted sound has fewer higher-frequency components than an air-conducted sound. The decay of transfer characteristics is approximately −10 dB/oct in the audible range, which is roughly parallel to the spectral characteristics of the measured NAM. [Work supported by SCOPE-S.]

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Session 5aSCb

Speech Communication: Production I (Poster Session)

Dani M. Byrd, Cochair
Univ. of Southern California, Linguistics, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693

Kiyoshi Honda, Cochair
ATR Human Information Processing Labs., 2-2 Hikaridai, Seika-Cho Soraku-Gun, Kyoto 619-0288, Japan

Contributed Papers

All posters will be on display and all contributors will be at their posters from 11:00 a.m. to 12:00 noon.

5aSCb1. Computer graphics facial animation of Japanese speech using three-dimensional dynamic visemes. Makoto J. Hirayama (Kanazawa Inst. of Technol., 7-1 Ohgigaoka, Nonoichi Ishikawa, 921-8501 Japan, mako@infor.kanazawa-it.ac.jp)

Speech production makes not only acoustic signals but also visual images of the face especially for the jaw, lips, teeth, and tongue. To reproduce realistic facial images during Japanese speech, a three-dimensional computer graphics model of Japanese visemes was made. These visemes were extracted from the video database of images around lips during speech captured by two high speed (up to 300 frames per second) cameras [M. J. Hirayama, Proc. ICPhS 2003 (2003), pp. 3157–3161]. Most of the visemes are created as static shapes. They are for five vowels, semi-vowels, and some consonants. For explosives by labials (/p/ /b/ /n/) or tongue (/t/ /d/ /n/), dynamic information, that is, multiple shapes and timing information were assigned for each viseme. By placing these visemes onto a time axis at the timing of phonemes of a sentence, then interpolating shapes in between by using a spline interpolation technique on speech articulators' motion graphs, computer graphics animation was made by ray-tracing rendering. [A part of this work was supported by Japan MEXT Academic Frontier Project.]

5aSCb2. A photoglottographic (PGG) method with external lighting and sensing. Kiyoshi Honda (LPP-UMR7019, CNRS/Univ. Paris 3, Paris F75005 France, and ATR, Kyoto 619-0288, Japan, honda@atr.jp), Shinji Maeda (CNRS/ENST, Paris, F75634 Cedex13 Paris, France), Stephane Hans (Service ORL, F75015 Paris, France), and Lise Crevier-Buchman (CNRS/Univ. Paris 3, Paris F75005 France)

Photoglottography (PGG) is a moderately invasive technique to observe glottal movements and vocal-fold vibrations. A light guide is inserted into the pharynx and transglottal light is detected by a photosensor placed on the outer skin below the glottis, or conversely the sensor is placed inside the pharynx and the subglottal cavity is illuminated from outside with a light source. This report evaluates a noninvasive PGG method, in which the hypopharynx is illuminated with a light guide placed on the neck skin at the level between the hyoid bone and thyroid cartilage and a photosensor on the skin near the cricoid cartilage. The light passed through the skin illuminates the lumen of the hypopharynx and the photosensor detects indirect (diffusion) light through the glottis during glottal adduction. Choosing an appropriate location for the light source and photosensor, glottal adduction pattern is observed for voiceless stops and fricatives in low as well as high vowel context. Thus, our diffusion-light PGG potentially overcomes the tongue-position dependency of the detected signals, which has been a major caveat of the conventional PGG methods.

5aSCb3. Evaluation of a photonasography. Angélique Amelot (LPP, 19 rue des Bernardins 75005 Paris, angelique.amelot@univ-paris3.fr), Kiyoshi Honda, Shinji Maeda (ENST, 75013 Paris, France), Lise Crevier-Buchman (HEGP, 75015 Paris, France), and Patricia Basset (LPP, 75005 Paris, France)

A photonasographic (PNG) technique was tested to evaluate velopharyngeal port (VP) opening/closing during speech. This system uses a photodetector placed in the pharynx via a nostril and a fiberscope in another nostril pointing to the back wall of the nasopharynx. With this setting, the light is reflected from the walls, passes through the open port, and is sensed by the photodetector. In a pilot experiment, a native French speaker read sequences C1V1C1 V2C1V1, where C1 = /n/ or /t/ and V1,2 = /a/, /i/, /A/, /E/, or /O/, as /tatA ta/. The obtained PNG data compare qualitatively well with the velum position data, on the same corpus, derived from the naso-fiberscopic video images. When NPG detects a VP opening, the fiberscope data correspondingly indicate a lowering of the velum and inversely. PNG, as well as the velum position data, show that VP is open at the onset of a nasal phoneme and is closed just after the offset of that nasal phoneme. Moreover, VP is not completely closed during the first and second phoneme in the sequence [ninini] and [nanana], indicating their nasalization. The combination of a well-set fiberscopic light-source and photodetector appears to give a good estimation of the degrees of VP opening.

5aSCb4. An MRI-based 3-D tongue atlas for tongue modeling. Chao-Min Wu and Han Tiet Goh (Dept. of Elec. Eng., Natl. Central Univ., 300, Chun-Da Rd., Chung-Li, Taiwan 320)

The main objective of this study is to build an MRI-based 3-D tongue atlas with an established available spatial transformation technique. The subjects for the MRI data were eight male and eight female college students (19–28 years old) who are native speakers of mandarin with Taiwanese accent without speech disorders. The oral MR images (axial: TR, 400 ms; TE, 10 ms, FOV 24×24; image matrix, 256×256 for 35 slices with 2-mm thickness) were acquired using a GE SIGNA 1.5-T scanner in the University Hospital of Chung Shan Medical University. The axial MR images of the human tongue were first segmented with snakes active contour method, then the 3-D tongues of each subject were reconstructed with morphology-based gray-level interpolation. Finally, these 3-D tongues were spatial transformed into a 3-D tongue atlas with thin-plate spline method. The 3-D tongue atlases for male and female subjects and mor-

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phometric studies, such as subject-to-subject, subject-to-atlas, and male-to-female, of these two atlases will be presented. Implication of these data related to tongue modeling will also be discussed. These information will be used as the basis for future research on tongue modeling and underlying physiological mechanism of normal and disordered speech production.

5aSCb5. Direct measurement of glottal volume velocity using high-speed, stereoscopic, particle imaging velocimetry. David A. Berry, Juergen Neubauer, and Zhaoyan Zhang (The Laryngeal Dynam. Lab., UCLA Div. of Head and Neck Surgery, 1000 Veteran Ave., Rm. 31-24, Los Angeles, CA 90095-1794)

Despite its central importance as the glottal source in several long-standing models of voice production, the glottal volume velocity signal has been relatively little studied. In particular, few direct measurements of the glottal volume velocity signal have been made. One common method for estimating the glottal volume velocity signal has been linear inverse filtering of the oral acoustic output. However, a major assumption of linear inverse filtering, and of the linear source-filter theory of speech production in general, is that the glottal source and the vocal tract are independent of each other. The accuracy of this assumption has been questioned, particularly at extremes of fundamental frequency and intensity. In order to test the limits of linear inverse filtering, direct, instantaneous measurements of glottal volume velocity have been made using high-speed, stereoscopic, particle imaging velocity on a self-oscillating, physical model of vocal-fold vibration. These direct measurements are contrasted with results from linear inverse filtering.

5aSCb6. Lip protrusion and tongue position in French vowels produced by blind speakers and sighted speakers. Lucie Menard, Annie Leclerc, Jerome Aubin, and Annie Brasseur (Phonet. Lab., UQAM, Case Postale 8888, succ. Ctr.-Ville, Montreal, H3C 3P8, Canada)

The influence of visual experience in speech production was investigated through a study of the French vowels /i y u a/. Articulatory and acoustic recordings of ten repetitions of the vowels in CVC syllables embedded in carrier sentences were conducted in three consonant environments (/b d g/) and two prosodic conditions (neutral and under contrastive emphasis). Six congenitally blind adults and six sighted adults were recorded. All participants were native speakers of Canadian French and had no history of speech disorder. The audio signal, lip movement, and tongue shapes were recorded using a digital camera and an ultrasound system. The minimal (for /i a/) or maximal (for /y u/) horizontal position of the upper lip was tracked and the front-back position of the tongue was measured for each vowel. Formant frequencies were extracted using LPC analyses. Results show that, despite similar acoustic differences, the rounded and unrounded vowels are less differentiated along the protrusion dimension for blind speakers compared to sighted speakers. It is suggested that this variability is related to a trade-off between lip protrusion and tongue position. Significant interaction effects between consonant environment and vowel category are discussed for each speaker group. [Work supported by SSHRC, NSERC.]

5aSCb7. Tongue muscle fiber tracking during tongue protrusion and rest. Hideo Shinagawa, Emi Z. Murano (Dept. of Biomed. Sci., Univ. of Maryland Dental School, 666 W. Baltimore St., Baltimore, MD 21201, hshinagawa@umaryland.edu), Jiachen Zhuo, Rao P. Gullapalli (Univ. of Maryland Medical Ctr., Baltimore, MD 21201), Bennett Landman, Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD 21218), and Maureen Stone (Univ. of Maryland Dental School, Baltimore, MD 21201)

Diffusion tensor imaging (DTI) has recently become a promising tool to investigate not only the nerve structure of the brain, but also integration of the peripheral muscle structure. The purpose of this study was to represent the functional deformation of tongue muscles with DTI by examining two conditions, tongue protrusion and rest. Three normal volunteers participated in the study. They were recorded at rest and wearing a custom made tongue retaining device (TRD), often used for patients with sleep apnea. The TRD held the tongue immobile in protruded position. In both positions, the tongue muscle structure and functional deformation could be visualized. In particular, the deformation of the genioglossus (GG) muscle, which is generally fan shaped, was clearly visible. The fibers of GG anterior muscle were more vertical with the TRD. The DTI technique has the potential to reveal not only myoarchitecture but also functional deformation of the tongue *in vivo*. With a little ingenuity, it might be possible to conquer the MRI sensitivity to small motion artifacts or improve tongue muscle stabilization to track the muscle fiber architecture during sustained speech. [Work supported by JSPS and NIH.]

5aSCb8. The effects of the false vocal folds on laryngeal pressures and flows. Li Sheng (Dept. of Biomed. Eng., School of Life Sci. and Technol., Xi'an Jiaotong Univ., Xi'an 710049), Ronald C. Scherer (Bowling Green State Univ., Bowling Green, OH 43403), Wan MingXi, Wang SuPin, and Qi LiYun (Xi'an Jiaotong Univ., Xi'an 710049)

As a laryngeal constriction, the false vocal fold (FVF) gap (the distance between the medial edges of the FVFs) may create important effects on phonation by altering the pressures within and flows through the larynx. The computational code FLUENT was used to examine the effects on pressures and flows for FVF gaps ranging from 0.02 to 2.06 cm, for three glottal angles (uniform and convergent/divergent 40 deg) and two minimal glottal diameters (0.04 and 0.06 cm), for constant subglottal pressure (8 cm H_2O). The specific design of the FVFs followed prior literature. Results suggest three important ranges of FVF gaps: (1) when the FVF gap was 1.5–2 times the minimal glottal diameter, pressures were lower throughout the larynx, and flows were higher through the larynx (less flow resistance); (2) for smaller FVF gaps, intralaryngeal pressures increased and flows decreased; (3) for greater FVF gaps, pressures and flows were unaffected. Also, (4) the divergent glottal angle gave the greatest flows, (5) flow separation locations for the divergent glottis moved downstream as the FVF gap decreased, and (6) the FVFs straightened the flow for narrow gaps. Thus, the status of the FVFs needs to be considered for voice production and synthesis.

5aSCb9. Acoustic and aerodynamic effects of false folds and epiglottis. Fariborz Alipour, Sanyukta Jaiswal, and Eileen Finnegan (Dept. Speech Path. and Audiol., Univ. of Iowa, Iowa City, IA 52242)

Excised larynx research usually excludes the false folds in an effort to reveal the true vocal folds. The purpose of this study was to examine the aerodynamic and acoustic effects of the false folds and epiglottis on excised larynx phonation. Canine larynges were mounted over a tapered tube on the excised larynx bench that supplied pressurized, heated, and humidified air. Glottal adduction was accomplished by passing sutures that simulated lateral cricoarytenoid muscle activation. First, the excised larynx with intact false folds and epiglottis was subjected to a series of pressure-flow experiments with tension and adduction as major control parameters. Then, the epiglottis and false folds were successively removed and the experiment was repeated for each condition. The subglottal pressure, EGG, flow rate, audio signal, and sound pressure level were recorded during each experiment. Glottal flow resistance was calculated from the pressure and flow signals, while the EGG signal was used to extract fundamental frequency. It was found that these additional structures have positive contribution to the glottal resistance and sound intensity of the larynx. Also, vocal fold elongation and glottal medial compression caused an increase in the glottal resistance. [Work supported by NIDCD Grant No. DC03566.]

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Session 5aUW

Underwater Acoustics and Acoustical Oceanography: Geoacoustic Inversion

Dezhang Chu, Cochair
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Chair's Introduction—7:45

Contributed Papers

7:50

5aUW1. Notes on broadband seabed geoacoustic inversion in shallow water. Ji-Xun Zhou and Xue-Zhen Zhang (Georgia Inst. of Technol., Atlanta, GA 30332-0405 and Inst. of Acoust., Chinese Acad. of Sci., Beijing, 100080, China)

With increasing interest in shallow-water environments, where bottom interaction is a dominant effect, there is a need to acquire accurate information about seabed geoacoustic parameters. Due to the difficulties and high costs of directly measuring seabed parameters at low frequencies (LF) in shallow water (SW), seabed geoacoustic inversion from long-range propagation has become an active research area. Much progress on this subject, notably on inversion methods, has been made in the last two decades, mainly through the use of powerful numerical codes and data processing tools. Despite this, seabed geoacoustic inversion often fails to yield systematic LF seabed parameters in a broad band (such as speed/attenuation vs frequency) that are required for seabed geoacoustic modeling or for sonar performance prediction in the SW environment. This paper will discuss several basic concepts and technical issues related to SW geoacoustic inversion, including filter phase shifting, speed-attenuation coupling, hidden depth, and sea-surface effects. It shows that inverting the broadband seabed acoustic parameters is a delicate task that can often be subject to errors. These issues should be considered in designing experiments that test seabed geoacoustic inversion methods or in interpreting experimental results. [Work supported by ONR and NNSF of China.]

8:05

5aUW2. Efficient use of *a priori* data in sediment inversions through the use of null space. Gopu R. Potty, James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882), Ying-Tsong Lin, and James F. Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA)

We present improved inversions for sediment parameters by incorporating *a priori* information about the environment. This improvement is achieved by projecting a desirable solution into the null space of the inversion and including this null space contribution along with the standard non-null space contribution. We use singular value decomposition (SVD) to define the null space of the inversion and elucidate our projection method. The desirable solution, which is projected into the null space, is constructed based on previous data from cores, geophysical surveys, and historic data. This approach introduces user bias into the solution; the projection onto the null space supplies a safety net by showing which aspects of this bias are justified by the data. The user bias can be considered as additional data which, when incorporated, can lead to meaningful solutions. Effectiveness of probing the null space will be compared to the more conventional nonlinear inversion schemes. This approach will be tested using field data collected as part of the Shelfbreak Primer experiment. [Work supported by ONR.]

8:20

5aUW3. Robust source localization and geoacoustic inversion in the Haro Strait Primer. Rashi Jain, Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ 07102), and Alex Tolstoy (A. Tolstoy Sci., McLean, VA 22101)

Gibbs sampling, a Markov chain Monte Carlo technique, has been shown to be apowerful tool for geoacoustic inversion and source localization. By providing estimates of posterior joint distributions, it offers a global optimization route for multidimensional estimation that reports uncertainty and covariance in addition to point estimates. In this work, Gibbs sampling is applied for extracting time delays from recorded time series during the Haro Strait primer experiment. Employing time delay estimates and using a linear approximation to the inverse problem and then regularization, estimates are obtained for source and receiver location and some environmental parameters. Simultaneously processing receptions at all three vertical line arrays for localization of each source reduces ambiguities in the estimation process. Similarly, using received signals corresponding to several sources, array element localization for a single array, a difficult problem for this particular data set, becomes more precise. Multiple data sets are used in the inversion and consistent results validate the robustness of the approach. Estimated bathymetry is in agreement with bathymetric maps for the region. [Work supported by ONR.]

8:35

5aUW4. Coherent noise processing and geoacoustic inversion. Peter Gerstoft, Chen-Fen Huang, and William S. Hodgkiss (Marine Physical Lab., Univ. of California San Diego, San Diego, CA 92093-0238)

Ocean acoustic noise can efficiently be processed to extract Green function information from noise [Roux *et al.*, J. Acoust. Soc. Am. (2004), Siderius *et al.*, *ibid.* (2006)]. By cross-correlating the ambient noise field from two sensors, it is possible to extract the impulse response between the two sensors including bottom and subbottom bounces. When this noise processing is used on a vertical array, it can give valuable information about the subbottom near the array. This information will be then be used to constrain a classical geoacoustic inversion procedure where we use a distant towed source to obtain the geoacoustic bottom parameters.

8:50

5aUW5. Geoacoustic inversion based on both acoustic pressure and particle velocity. A. Vincent van Leijen (NLDA, P.O. Box 10.000, 1780 CA, Den Helder, The Netherlands, av.vanleijen@kim.nl), Jean-Pierre Hermand (ULB, B-1050 Bruxelles, Belgium), and Kevin B. Smith (NPS, Monterey, CA 93943)

Conventional inversion schemes for environmental assessment depend on an objective function that exploits amplitude or phase information of acoustic pressure data alone. This work investigates the potential of vector sensors for geoacoustic inversion by defining an objective function that

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also takes into account acoustic particle motion. Calculations are performed on synthetic broadband data for a shallow water environment (South Elba) with an optimization scheme based on different metaheuristics. Differences in the inversion process, including sensitivity of the cost function to environmental parameters and convergence speed of the optimization algorithm, are presented by comparing inversion results for a sparse pressure-only array and a vector sensor array.

9:05

5aUW6. Geophysical parameter inversion in a range-dependent environment. Woojae Seong, Keunhwa Lee, Kyungsup Kim (Dept. of Ocean Eng., Seoul Natl. Univ., Seoul, 151-742, Korea), and Seongil Kim (Agency for Defense Development, Jinhae, Korea)

Matched-field inversion technique is applied for estimation of geophysical parameters of the ocean bottom in a range-dependent shallow water. In the experiment (MAPLE-4), conducted off the coast of the East Sea during May 2005, narrow-band multitone cw acoustic data were obtained from the towed moving source along a weakly range-dependent track, from 2 to 18 km apart from the L-shaped receiver array. In the inversion, complex density model based on Biot model is used to invert for parameters including porosity and permeability. Inversion results are compared with existing geological survey data. In addition, the effect of range dependency resulting from the seafloor slope and the existing bottom intrusion is examined.

9:20

5aUW7. Bayesian inversion of propagation and reverberation data. Peter L. Nielsen (NATO Undersea Res. Ctr., Viale S. Bartolomeo 400, 19138 La Spezia, Italy) and Stan E. Dosso (Univ. of Victoria, Victoria, BC, Canada V8W 3P6)

A Bayesian matched-field inversion approach to infer geoacoustic and scattering properties of the seabed is applied to simulated propagation and reverberation data received on a towed horizontal array. The approach is based on the method of fast Gibbs sampling (FGS) of the posterior probability density to estimate uncertainties in both geoacoustic and scattering parameters for broadband acoustic data in realistic shallow-water environments. The FGS is linked to an acoustic propagation model that simultaneously provides complex acoustic pressure at short propagation ranges and long-range reverberation intensity. The inversion algorithm is initially applied to long-range reverberation data alone to assess the geoacoustic information content of reverberation in terms of marginal posteriori probability densities for the environmental parameters. A reduction in uncertainty for the extracted geoacoustic and scattering parameters is demonstrated by a simultaneous inversion of the propagation and reverberation horizontal array data.

9:35

5aUW8. On the use of acoustic particle velocity fields in adjoint-based inversion. Matthias Meyer, Jean-Pierre Hermand (Université libre de Bruxelles, Belgium & Royal Netherlands Naval College, The Netherlands), and Kevin B. Smith (Naval Postgraduate School, Monterey, CA)

Following the recent interest in the use of combined pressure and particle motion sensors in underwater acoustics and signal processing, some general aspects regarding the modeling and multipath phenomenology of acoustic particle velocity fields in shallow water environments have been studied. In this paper we will address a number of issues associated with the incorporation of vector sensor data (pressure and particle velocity) into adjoint-based inversion schemes. Specifically, we will discuss the ability of a semi-automatic adjoint approach to compute the necessary gradient information without the need for an analytic model of the adjoint particle velocity field. Solutions to the forward propagation of acoustic pressure are computed using an implicit finite-difference parabolic equation solver while the particle velocity is calculated locally at each grid point. Some numerical examples of vector sensor inversion results are provided. [Work supported by Royal Netherlands Navy.]

9:50

5aUW9. Sensitivity analysis of a geoacoustic parametrized model and its application to inversion of seabed properties. Jin-Yuan Liu and Chung-Ray Chu (No. 70 Lien-hai Rd. Kaohsiung 804, Taiwan)

The work first aims to analyze the parametrized geoacoustic model proposed by Robins [J. Acoust. Soc. Am. **89**, 1686–1696 (1991)], in which the density and sound speed distributions vary with respect to depth as a generalized-exponential and an inverse-square function, respectively. The model contains a set of parameters that, by appropriate selection, may fit well the realistic geoacoustic variations [E. L. Hamilton, J. Acoust. Soc. Am. **68**, 1313–1340 (1990)]. By choosing the plane-wave reflection field as an objective function, each model parameter is carefully analyzed to determine its range and sensitivity. Then, numerical simulation is employed to establish an inversion procedure, in conjunction with the application of acoustic wave reflection from a nonuniform seabed. Finally, a field experiment is designed and implemented to estimate the seabed acoustic properties based upon the model parameter inversion.

10:05–10:20 Break

10:20

5aUW10. Characterization of sediment dynamics in an estuary environment using acoustic techniques. Jean-Pierre Hermand, Laura Perichon (Environ. Hydroacoustics Lab., Optics and Acoust. Dept., ULB-CP 194/05, 50 AV. F.D. Roosevelt, B-1050 Brussels, Belgium), and Michel Verbanck (Univ. Libre de Bruxelles, B-1050 Brussels, Belgium)

In recent years, acoustic-based methods have been developed to characterize the dynamical behavior of loose sediments and bed deposits in very shallow water environments. In this paper, we present preliminary results on the estimation of the dynamic changes in an estuarine environment using data from dual-frequency echosounding at high resolution and contemporaneous hydrological measurements including suspended matter concentration, density subbottom profiling, and data assimilation based on a sediment transport model. Those measurements are being conducted in the lower estuary of the Scheldt (Belgium) at the Sint Anna site where strong tide and season-dependent phenomena can be observed. This allows us to construct a ground-truthed, time-dependent geoacoustic model of the environment, i.e., a characterization of sound speed, density, and attenuation in function of time and depth. Synthetic acoustic data generated by that model will then be used to test inversion methods for monitoring sediment dynamics in real time.

10:35

5aUW11. Geoacoustic model for the New Jersey Shelf by inverting airgun data. Yong-Min Jiang, N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3055, Victoria, BC V8W 3P6, Canada), and Mohsen Badiey (Univ. of Delaware, Newark, NJ 19716)

This paper describes geoacoustic inversion of airgun data acquired during the SWARM95 experiment. Hybrid optimization and Bayesian inversion techniques were applied to three airgun data sets recorded by a vertical line array. Optimization results are used to show the consistency of the estimates from all of the shots in terms of histograms and standard deviations of the inverted geoacoustic model parameters. The inversion results from the Bayesian approach are used to show the uncertainties of the estimates in terms of marginal distributions, MAP estimates, and credibility intervals. In the Bayesian inversion, full data error covariance matrices were estimated by ensemble averaging the covariance of the residuals of the measured and modeled data of inversions from many shots. The numbers of shots in the ensemble averages were determined by checking the temporal coherence of the signal. Statistical tests were used to test the validity of the assumptions in the Bayesian approach after incorporating full data error covariance matrices. With these inversion techniques, equivalent geoacoustic models with/without shear wave estimates are extracted for this experimental site. The frequency dependence of the p-wave attenuation, and the correlation between the geoacoustic parameters are obtained from the inversion results. [Work supported by ONR.]

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10:50

5aUW12. Mismatch effects on passive matched-field, thick-sediment inversion. Donald R. DelBalzo, Robert S. Linzell, and Lisa A. Pflug (Planning Systems Inc., 40201 Hwy. 190 E, Slidell, LA 70461)

Acoustic field predictions in shallow-water areas can be severely limited by environmental uncertainty, especially in sediment property characteristics. Direct methods of obtaining geophysical data are expensive, so inverse methods, which invert signals from controlled active sources, have recently become popular. This work assesses the impact of environmental mismatch on low-frequency (<100 Hz) matched-field correlations on broadband signals from surface ships with unknown source levels at unknown ranges. A range-staggered technique is employed to create a range-dependent sediment description, with ever-increasing confidence as additional ships are analyzed. Matched-phase techniques are applied in a simulated shallow-water environment with a single vertical array and high signal-to-noise ratios. The simulations indicate significant potential for passive, range-dependent, thick-sediment characterizations out to ranges of tens of water depths in shallow water, despite reasonable mismatch conditions in the water and in the sediments. [Work sponsored by SPAWAR.]

11:05

5aUW13. To obtain geo-acoustic parameters by the acoustic inversion technique using subbottom profiler. Chung-Wu Wang and Cheng-Che Lee (China College of Marine Technol. and Commerce, No. 212, Sec. 9, Yan-Ping N. Rd.,Taipei, Taiwan, ROC)

The fast sound speed at sea surface in the water surrounding Taiwan causes strong refraction. If the water depth is shallower than 5000 m, the sound ray will reflect at the bottom, causing severe bottom losses. Therefore, geo-acoustic parameters are important for acoustic studies in the water surrounding Taiwan. In the past, researchers used single-pulse sonar of 3.5 kHz to obtain geo-acoustic parameters by the spectral ratio technique. Now the sediment exploration has already advanced and used the Chirp Sonar system on Taiwanese Research Vessels such as Ocean Researcher I (OR I) and Ta-Kuans TOPAS system. This paper presents the results of attenuation coefficients in ocean bottoms using the acoustic inversion technique on the chirp sonar data collected by OR I. The results are compared with the core data, and the error is within 15%.

11:20

5aUW14. The resolution of a nonlinear ocean acoustic inverse problem. Robert I. Odom and Andrew A. Ganse (Appl. Phys. Lab. and Dept. of Earth and Space Sci., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698)

For a strictly linear inverse problem the model resolution matrix is a simple product of the linear system matrix and its generalized inverse. How much the model resolution differs from the identity is an indication of bias in the inverted model. Nonlinear inverse problems lead to higher order resolution operators, which allow one to quantify the effect of ignoring the nonlinearity in the inverse problem. In addition, the trade-off between resolution and variance is examined for the nonlinear inverse problem. Ocean acoustic inverse problems for ocean and/or bottom structure are intrinsically underdetermined, because some continuous quantity such as sound speed or temperature is estimated from a finite number of data values, and the model is generally discretely parametrized to be numerically tractable. Discretizing the model has the positive effect of regularizing an intrinsically ill-posed problem. However, discretizing an intrinsically nonlinear problem extracts a penalty. Higher order resolution operators may contribute even when they should be identically zero. (For example, in a quadratically nonlinear model, cubic and higher order resolution operators should be zero, but won't be zero as a result of the discretization.) Modal propagation in an ocean acoustic waveguide is used to illustrate the issues. [Work supported by ONR.]

11:35

5aUW15. Adapting results in filtering theory to inverse theory, to address the statistics of nonlinear geoacoustic inverse problems. Andrew A. Ganse and Robert I. Odom (Appl. Phys. Lab. and Dept of Earth and Space Sci., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

The intrinsically non-Gaussian statistics of nonlinear inverse problems, including ocean geoacoustic problems, is explored via analytic rather than numerical means. While Monte Carlo Bayesian methods do address the non-Gaussian statistics in nonlinear inverse problems, they can be very slow, and intuitive interpretation of the results are at times problematic. There is great theoretical overlap between recursive filters/smoothers, such as the extended Kalman filter, and methods of linear and nonlinear geophysical inversion. The use of recursive filters in inversion is not in itself new, but our interest is in adapting statistical developments from one to the other. Classic analytic methods in both filtering theory and inverse theory assume Gaussian probability distributions, but newer nonlinear filters do not all make this assumption and are explored for their potential application to nonlinear inverse problems. The similarities and differences between the frameworks of filtering theory and inverse theory are laid out in a series of geoacoustic inverse problem examples. Recent work in nonlinear filters handles non-Gaussian probability densities that are constrained to a particular form, and also derives analytic expressions for higher order moments of these density functions. The application of these developments to geoacoustic inverse problems is addressed. [Work supported by ONR.]

5a SAT. AM

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Session 5pAA

Architectural Acoustics, Speech Communication and Psychological and Physiological Acoustics: Psychological Aspects of Speech in Rooms II

John S. Bradley, Cochair
National Research Council, Inst. for Research in Construction, Ottawa, Ontario K1A 0R6, Canada

Hiroshi Sato, Cochair
National Inst. for Advanced Industrial Science Technology, Inst. for Human Science and Biomedical Engineering, 1-1-1 Higashi, Tsukuba, Ibaraki 305-8566 Japan

Contributed Papers

1:00

5pAA1. Calibration of consonant perception in room reverberation. Kanako Ueno (Inst. of Industrial Sci., Univ. of Tokyo, 4-6-1 Komaba, Meguro-ku, Tokyo, 153-8505, Japan), Norbert Kopco, and Barbara Shinn-Cunningham (Boston Univ., 677 Beacon St., Boston, MA 02215)

Many studies of sound perception often assumed that our auditory sensory processes are relatively static, rather than plastic. However, in everyday environments, we naturally and fluidly compensate for interfering effects of background noise and room reverberation. In order to investigate how listeners calibrate auditory perception to such acoustic interference, a listening experiment was performed to measure the effect of sudden changes of reverberation on the identification of consonants. Test sounds were generated by convolving two types of binaural room impulse responses (BRIRs) measured in large real rooms with speech tokens. As a control condition, pseudo-anechoic BRIRs with negligible reverberation energy were used. Listeners were asked to identify the consonant in a vowel-consonant target. The target was preceded by a carrier phrase consisting of vowel-consonant pairs from the same talker. In some cases, the target and carrier phrase were processed by the same BRIRs, while in others the BRIR's processing target and carrier differed. Consistent effect of calibration was observed in one of the simulated rooms, but not in the other, suggesting that the ability to compensate for the effects of reverberation depends on the specific pattern of reverberation produced in a given room. [Work supported by AFOSR and NSF.]

1:15

5pAA2. Sentence context influences vowel perception in reverberant conditions. Janine Wotton (Dept. of Psych., Gustavus Adolphus College, 800 W. College Ave., St. Peter, MN 56082, jwotton2@gac.edu), Kristin Welsh, Crystal Smith, Rachel Elvebak, Samantha Haseltine (Gustavus Adolphus College, St. Peter, MN 56082), and Barbara Shinn-Cunningham (Boston Univ., Boston, MA 02115)

Sentences recorded with a Mid-western accent were convolved with head-related impulse responses that included different room reverberation conditions. The stimuli were presented binaurally through headphones in an echo-attenuated chamber and subjects ($n=23$) typed the sentences they heard. The target word was one of a vowel pair (cattle/kettle, jam/gem, gas/guess, past/pest) embedded as the second word in one of three sentence types. The neutral sentence provided little context for the word. Target words in sentences that provided strong contextual cues could be congruent or incongruent with the expectations of the subject, for example, "The cattle/kettle grazed in the meadow." Subjects made significantly more errors in the incongruent sentences compared to the neutral (Wilcoxon$=3.572$ $p<0.05$) or congruent sentences (Wilcoxon$=3.56$ $p<0.05$). When the target word was in a congruent sentence subjects performed equally well in reverberant or pseudo-anechoic conditions (Wilcoxon$=1.298$) but they made more errors in the reverberant condition for both neutral (Wilcoxon$=3.359$, $p<0.05$) and incongruent sentences (Wilcoxon$=2.241$, $p<0.05$). Results suggest that reverberation may cause listeners to rely more heavily on linguistic context to determine word meaning. [Work supported by NOHR, AFOSR.]

1:30

5pAA3. Perceptual compensation for reverberation: Effects of noise-context bandwidth. Simon J. Makin, Anthony J. Watkins, and Andrew P. Raimond (School of Psych., The Univ. of Reading, Earley Gate, Reading RG6 6AL, UK, s.j.makin@reading.ac.uk)

Perceptual compensation for reverberation is observed when the reverberation is applied to a test word (from a "sir"-to-"stir" continuum) and its context (e.g., "next you'll get_to click on") are varied independently. Increasing reverberation in test words decreases listeners' "stir" responses, as reverberation "fills the gap" that cues the [t]. Compensation occurs when the context's reverberation is commensurately increased, and "stir" responses increase back to the level found with minimal test-word reverberation. Compensation is strongest with speech contexts but also occurs with some noise-like contexts, including "signal-correlated noise" that has the wideband temporal envelope of the original speech. Also effective is a wideband noise that is given the temporal envelope seen at the output of a single auditory filter in response to speech. A narrow-band version of this "auditory-filtered" noise is not effective, but when contexts are made by summing of three or five of these bands, their effectiveness increases correspondingly. Compensation appears to be informed by the "tails" that reverberation adds at offsets, so it merely requires contexts with suitable temporal-envelope fluctuations. However, effects seem confined to the context's frequency region, so the crucial offsets need to be in a wide range of frequency bands. [Work supported by EPSRC.]

1:45

5pAA4. Aural localization of speech stimuli. Evelyn Way (Talaske, 105 N. Oak Park Ave, Oak Park, IL 60301, evelyn@talaske.com)

Localization error was tested for a variety of signals to answer the question: do humans aurally localize different speech stimuli with a different localization blur? A series of tests was conducted comparing the effect of the sentence length, gender of the talker, and frequency content of the signals on localization. Results were applied to ongoing research into constructing an aurally accurate telepresence system.

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2:00

5pAA5. Quantifying the effects of room acoustics on speech intelligibility in very low signal-to-noise ratio environments. Jarrod E. Whittington (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180, whittj@rpi.edu) and John S. Bradley (Inst. for Res. in Construction, Ottawa, K1A OR6 Canada)

The intelligibility of speech transmitted from closed offices to adjacent spaces is strongly affected by the signal-to-noise ratio at the receiver position and the acoustical characteristics of the spaces involved. Previous studies have suggested that the effect of room acoustics on speech intelligibility in closed offices and rooms is negligible and can be ignored (as with intelligibility quantifiers such as the articulation index). The purpose of this study is to show that in conditions of very low signal-to-noise (i.e., when high speech privacy is a primary concern), the influence of room acoustics rises dramatically. To this end, multiple subjects were given tests of speech intelligibility in simulated sound fields. Speech samples were presented to subjects with seven levels of signal-to-noise and four different reverberation times. The results from these tests show that as reverberation time increases, speech intelligibility decreases much more sharply for very low (-8 dB) signal-to-noise situations than in higher ($+10$ dB) signal-to-noise situations. This suggests an important relationship between room acoustics and speech privacy/security.

2:15

5pAA6. The effect of a preprocessing approach improving speech intelligibility in reverberation considering a public-address system and room acoustics. Nao Hodoshima, Takayuki Arai (Dept. of Elec. and Electron. Eng., Sophia Univ., Tokyo 102-8554, Japan), and Peter Svensson (Norwegian Univ. of Sci. and Technol., Trondheim NO-7491, Norway)

This study evaluates a preprocessing approach for reducing reverberation effects when the input signal is not ideal, dry speech, but rather a realistic speech signal picked up by a close microphone in a room. And this study shows how the situation affects the input and the further approach compared to a dry signal as the input. Steady-state suppression, as described by Arai *et al.* [Acoust. Sci. Technol. **23**, 229–232 (2002)], was used as a preprocessing approach that processes a speech signal before it is radiated from loudspeakers. A lecture was simulated in two different halls (reverberation times of 1.2 and 1.8 s) in which public address systems were installed. The simulation software CATT-Acoustic was used and impulse responses were calculated for the input to the preprocessing approach and for a listener position. Stimuli for a syllable identification test were prepared by convolving speech signals with the calculated impulse responses. Speech signals were given with and without steady-state suppression. The inclusion of natural and electroacoustical impulse responses makes the study of steady-state suppression more realistic and tests its robustness. [Work supported by JSPS (176911).]

2:30

5pAA7. Improving speech intelligibility in reverberant rooms. Douglas F. Winker and Elmer L. Hixson (Dept. of Elec. and Comp. Eng., Univ. of Texas, Austin, TX 78712)

Speech intelligibility in rooms with long reverberation times has long been a problem for acousticians. This problem is exacerbated when the option for acoustic treatments is not an option for a variety of reasons. In this study, a constant-beamwidth, wide-bandwidth (CBWB) loudspeaker array was used to improve speech intelligibility in two multipurpose rooms where treatment was not an option. A CBWB loudspeaker array was designed with independent beam pattern control on both the horizontal and vertical axes. A wide horizontal beam pattern and narrow vertical pattern were designed to achieve constant coverage from 500 Hz to 4 kHz. The array incorporates four separate arrays, two with nested elements, and linear-phase FIR filters to maintain constant beamwidth over the three octaves of interest. Room simulations were conducted with CATT-Acoustic and the array was implemented and compared to a source with a more conventional coverage pattern. The STI method was used to measure speech intelligibility at several positions throughout the rooms. The study showed an improvement in speech intelligibility and a reduction in measured reverberation times.

2:45

5pAA8. Experimental investigations of the influence of room acoustics on the teacher's voice. Malte Kob, Anja Kamprolf, Christiane Neuschaefer-Rube, Oliver Goldschmidt (Chair of Phoniatrics and Pedaudiology, RWTH Aachen Univ., Pauwelsstr. 30, 52074 Aachen, Germany, mkob@ukaachen.de), and Gottfried Behler (RWTH Aachen Univ., 52066 Aachen, Germany)

Teachers belong to the group of professional voice users who often suffer from voice disorders. One reason for a significantly increased prevalence of voice problems can be poor room acoustical conditions in the classrooms. In this study, four rather reverberant and loud classrooms in a primary school in Aachen were analyzed using measurements of the reverberation time, T_{20}, and the speech transmission index, STI. About half of the school's teachers were investigated with respect to their voice status by using phoniatric, logopedic, and objective voice analysis methods. The prevalence of voice problems in this group was found to exceed previous studies where subjective voice quality was rated. In a second part of this joint project the change of voice quality during the teachers' working day was analyzed. Two of the four rooms were acoustically optimized. Members of two groups of teachers with and without voice problems were recorded before and after teaching in either one of the acoustically poor rooms or one of the newly renovated rooms. The preliminary results indicate changes of the voice quality in most subjects with respect to one or more voice parameters. Further studies shall prove the significance of the room influence.

3:00–3:15 Break

3:15

5pAA9. Human voice phoneme directivity pattern measurements. Brian F. G. Katz, Fabien Prezat, and Christophe d'Alessandro (Percept. Située, LIMSI-CNRS, BP 133, F91403 Orsay, France)

The application of directivity patterns to radiating sources into computer simulations and auralizations is common for loudspeaker models. Few applications include the directivity patterns of natural sources, partly due to the lack of sufficient data. This work presents the results of a detailed measurement study on human voice directivity in three-dimensions. Unlike previous studies that have used average directivity data over read phrases, this work presents results that are measured for a number of sustained individual phonemes. Details of the measurement protocol and posttreatment processing are presented. Comparisons are made relative to phoneme, f0, spectral characteristics, and associated mouth geometry for several talker subjects. Studies have also been made on the directivity of the singing voice. Specifically, the variations in directivity relative to level (piano, fortissimo, etc.) and projection as controlled by the singer have been investigated. Results of this work are applicable to speech production research, talker simulator design, room acoustic sound field prediction, and virtual reality systems with talking avatars.

3:30

5pAA10. Case study for voice amplification in a highly absorptive conference room using a negative absorption tuning by the Yamaha Active Field Control system. Takayuki Watanabe (YAMAHA Commercial Audio Systems, Inc., 6600 Orangethorpe Ave., Buena Park, CA 90620)

The Yamaha Active Field Control (AFC) enhances and varies acoustical conditions from the room acoustics using the acoustical feedback in a system, which is properly tuned to the installed room acoustics. The system is primarily used to improve auditory impressions of reverberance, loudness by enhancement of reverberation in time, and energy in a music venue. Here, a negative absorption tuning is demonstrated as beneficial to

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the presentation room for a simple spoken speech used with a relatively excessive absorption at a middle-sized multiuse conference room. The space occupies 1100 ft^2 with a ceiling height of 9 ft. It is designed for demonstration of sound equipment as well as lecture without audio equipment. Since the acoustics in the room are highly absorptive (average absorption coefficient, 0.3), considering a primary use for product demonstration, it causes a flutter echo between longitudinal walls due to less sound energy in a late field. We demonstrate here a system adjustment of the AFC system emphasizing voice amplification as well as an acoustical solution to a defect of a fluttering echo here. Acoustical descriptors such as RT, echo diagram, and SPL are discussed for a tuned room acoustics with the AFC system.

Contributed Poster Papers

Poster papers 5pAA11 to 5pAA14 will be on display from 1:00 p.m. to 4:45 p.m. Authors will be at their posters from 3:45 p.m. to 4:45 p.m.

5pAA11. Slowed speech spreading into reverberant environments; steady-state suppression improves speech intelligibility. Yuki Nakata, Yoshiaki Murakami, Nao Hodoshima, and Takayuki Arai (Dept. of Elec. and Electrons Eng., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo, 102-8554 Japan)

Speech intelligibility is often decreased in reverberant environments. For improving speech intelligibility under reverberant environments, Arai *et al.* [Acoust. Sci. Technol. **23**, 229–232 (2002)] suggested steady-state suppression, which suppresses steady-state portions of speech to reduce overlap-masking (causing degradation intelligibility) and improves speech intelligibility as preprocessing. Previous studies showed that speech intelligibility was improved through this processing under some reverberant environments [Hodoshima *et al.*, J. Acoust. Soc. Am. **119**(6), 4055–4064 (2006)]. On the other hand, speech intelligibility is known to be increased greatly by speaking slowly [Bolt and MacDonald, J. Acoust. Soc. Am. **21**(6), 577–580 (1949)]. However, it is not optimal for improving speech intelligibility in terms of reducing overlap masking because vowels causing major parts of overlap masking are expanded by speaking slowly. Therefore, in the current study, we investigated the effects of steady-state suppression on speech perception with a decreased speech rate from 6 to 4 and 5 morae/s. Results showed that the slowest speech (4 morae/s) with steady-state suppression was the most intelligible. Also, steady-state suppression improved speech intelligibility at a speech rate of 4 morae/s in each reverberant condition (reverberation times of 1.5, 2.0, and 2.5 s). [Work partially supported by JSPS.KAKENHI (16203041).]

5pAA12. Suppression of speech intelligibility loss through a modulation transfer function-based speech dereverberation method. Masashi Unoki, Masato Toi, Yohei Shibano, and Masato Akagi (School of Information and Sci., JAIST, 1-1 Asahidai, Nomi, Ishikawa, 923-1292 Japan, unoki@jaist.ac.jp)

The concept of modulation transfer function (MTF) can successfully be applied to evaluate the quality of speech transmission in the room acoustics [Houtgast and Steeneken, J. Acoust. Soc. Am. **77**, 1069–1077 (1985)]. We previously proposed a speech dereverberation method based on the MTF concept, which consisted of MTF-based power envelope inverse filtering and the carrier regeneration in the filterbank [Unoki *et al.*, EuroSpeech2003 (2003)]. This paper evaluates how the proposed method can suppress the loss of speech intelligibility caused by reverberation, by comparing various methods. We have carried out massive simulations of dereverberation for reverberant speech signals to objectively evaluate these methods. We also subjectively evaluated the methods via listening tests. In these simulations, artificial reverberations were convolved with a clean speech signal in which the impulse responses in the room acoustics can be approximated from the exponential decay as a function of reverberation time with a white-noise carrier. The results of both evaluations show that, in addition to reducing the averaged log-spectrum distortion by about 1 dB, the proposed method reduces the loss of speech intelligibility by about 30%. [Work supported by a Grant-in-Aid for Science Research from the Japanese Ministry of Education (No. 18680017).]

5pAA13. Word and mora intelligibility in "Familiarity-controlled word-lists 2003 (FW03)." Tadahisa Kondo, Shigeaki Amano (NTT Commun. Sci. Labs., NTT Corp., 3-1 Morinosato-Wakamiya, Atsugi, Kanagawa, 2430198, Japan, tkondo@brl.ntt.co.jp), Shuichi Sakamoto, and Yôiti Suzuki (Tohoku Univ., Aoba-ku, Sendai, Miyagi, 9808577, Japan)

"Familiarity-controlled Word-lists (FW03)" was developed to make it possible to perform intelligibility tests on the same person repeatedly and/or under several different conditions. FW03 consists of 20 lists of 50 words in four word familiarity ranks. These words were selected taking phonetic balance into consideration to maximize their variety of initial moras and vowel-consonant sequences. To confirm that all FW03 word lists from a particular familiarity rank present the same hearing difficulty, we measured the word intelligibility of all the words in FW03. Sixteen subjects listened to 4000 words spoken by four speakers with seven signal-to-noise ratios. The mean intelligibility scores for the lists were significantly different even for the same familiarity rank. The intelligibility was apparently influenced by some specific moras and their position in words. These results suggest that it is difficult to equalize the word intelligibility of the lists even when the word familiarity is controlled more precisely. We therefore examined several methods for equalizing the intelligibility among the lists such as controlling the sound pressure level for each word when operating an actual intelligibility test. The effectiveness of these methods is discussed with detailed analyses of word and mora intelligibility in FW03.

5pAA14. Influence of Deutlichkeit value and reverberation time on improved speech intelligibility in reverberant environments because of steady-state suppression. Nahoko Hayashi, Nao Hodoshima, Takayuki Arai (Dept. of Elec. and Electron. Eng., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554 Japan, nhayash@sophia.ac.jp), and Kiyohiro Kurisu (TOA Corp., Takarazuka, Hyogo 665-0043 Japan.)

To improve speech intelligibility in reverberant environments, Arai *et al.* proposed "steady-state suppression (SSS)" as preprocessing [Arai *et al.*, Acoust. Sci. Technol. **23**, 229–232 (2002)]. In this study, a perceptual experiment under artificial reverberant conditions with simulated impulse responses was conducted to elucidate the effect of the Deutlichkeit (D) value and reverberation time (RT) on improvements of speech intelligibility because of SSS. Artificial impulse responses were simulated with white noise multiplied by a decay curve. The advantage of this method is that the simulated impulse responses have mutually similar frequency characteristics; consequently, we can evaluate them using only the D value and RT regardless of their different frequency characteristics. Two parameters, the energy of the impulse response 50 ms from the direct sound and the attenuation rate of the decay curve, were controlled to obtain several impulse responses having certain D value and RT. Results show that SSS improved speech intelligibility in the conditions of low D value, even if RT was long or short. We could also interpret these results as indicating that processing is effective when the original speech intelligibility is less than 60%. [Work supported by JSPS.KAKENHI (16203041).]

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Session 5pBB

Biomedical Ultrasound/Bioresponse to Vibration: Medical Ultrasound

Yuri A. Pishalnikov, Cochair
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Yoshiaki Watanabe, Cochair
Doshisha Univ., Faculty of Engineering, 1-3 Tatara-Miyakodami, Kyotamabe-shi, Kyoto 610-0321, Japan

Contributed Papers

1:00

5pBB1. Measurement of thrombi volume using a transesophageal echocoardiography movie. Masami Aoki (Grad. School of Sci. and Technol., Chiba Univ., 1-33 Yayoi-cho Inage-ku, Chiba, 263-8522, Japan), Tadashi Yamaguchi, and Hiroyuki Hachiya (Chiba Univ., Chiba, Japan)

In a total knee replacement operation, thrombi are formed by blood stasis in the deep venous of inferior limbs. They can cause pulmonary thromboembolism. The current judgment method of the existence of thrombi in the orthopedics field is visual observation using transesophageal echocoardiography (TEE). Therefore, we attempted to devise a quantitative measurement method of the volume of thrombi from TEE movies using multiple signal processing. It is necessary to extract a contour of the right atrium area at each frame of a TEE movie to measure the amount of thrombi that reach the lungs. Therefore, we propose processing that combines the modified brain extraction technique (BET) and vector moving method. The modified BET enables high-precision contour extraction. Vector moving performs the fast contour extraction. In addition, the fiber structure extraction technique (FSET) is used to extract the thrombi information in the right atrium. Similarly to speckle reduction, FSET can extract anomalous information from echo images. Using these processes, we observed a small periodic change that is synchronized with the heartbeat and the large time change that is synchronized with breathing.

1:15

5pBB2. Auscultation in high-noise environments using hybrid electromechanical and ultrasound-Doppler techniques. Adrianus J. M. Houtsma, Ian P. Curry (U.S. Army Aeromedical Res. Lab., Fort Rucker, AL 36362), John M. Sewell, and William N. Bernard (Active Signal Technologies, Inc., Linthicum Heights, MD 21090)

Medical evacuation of battlefield casualties or traffic accident victims by air typically takes place in very noisy environments. Auscultation of patients, e.g., to support chest intubation or to detect a pneumothorax, is therefore difficult or impossible to perform. A conventional acoustic stethoscope will not function very well in background noise levels beyond 80 dB. Electronic stethoscopes, in combination with mechanical impedance-matched transducer designs, can extend this range to about 90 dB, but this is not enough for helicopter noise levels that can reach 110 dB. The use of an ultrasound transmitter and receiver, however, provides an essentially noise-free auscultation channel since transportation vehicles do not produce acoustic energy at ultrasound carrier frequencies of 2–3 MHz. Clean and noise-free heart and breath sounds have been obtained in broadband noise fields of intensities as high as 120 dB. A hybrid stethoscope has been developed that allows auscultation by ultrasound-Doppler as well as electromechanical means. Pros and cons of making Doppler sounds subjectively similar to conventional sounds by nonlinear signal processing will be discussed, as well as potentially functional and meaningful aspects of Doppler signals that are not found in conventional stethoscope sounds.

1:30

5pBB3. Color and pulsed Doppler signatures of vascular bleeding. Wenbo Luo, Vesna Zderic, and Shahram Vaezy (Dept. of Bioengineering, Univ. of Washington, Seattle, WA 98195)

To develop ultrasound-based methods for detection and localization of internal vascular bleeding, we investigated the bleeding signatures using color and pulsed Doppler. Under ultrasound guidance, femoral arteries ($n=13$) and veins ($n=12$) in seven pigs were catheterized with 6, 9, and 11 F catheters. Color and pulsed Doppler were acquired from the injury site and the surrounding positions after withdrawal of the catheter. The bleeding rates were 3.8–38 ml/s. Color Doppler revealed that the injured vessels were in spasm. The checkered color pattern at the injury site indicated flow turbulence. Pulsed Doppler spectrum of the color jet (extravasated blood) represented a unique arterial bleeding pattern with increased baseline velocity (from 0 up to 25.8 cm/s) and elevated systolic velocity (from 23.2 up to 62.5 cm/s) at the puncture site, as compared to the normal arterial pattern. Venous bleeding showed a narrow-band vibration signal at the bleeding site. Local blood turbulence contained multiple velocity components which were illustrated as checkered color in color Doppler. Pulsed Doppler spectrum showed increased baseline due to turbulence and elevated velocity due to narrowed vessel. In future studies pulsed Doppler radio-frequency data will be used to analyze bleeding patterns for detection and localization of internal bleeding.

1:45

5pBB4. Basic study on angular dependence of ultrasonic scattering from wire phantom mimicking myocardium. Teppei Onodera, Hideyuki Hasegawa, and Hiroshi Kanai (Grad. School of Eng., Tohoku Univ., Sendai 980-8579, Japan, onodera@us.ecei.tohoku.ac.jp)

It is reported that the ultrasonic scattering from myocardium varies periodically during a cardiac cycle. One of the reasons is considered to be the change of the angle between the ultrasonic beam and the direction of myocardial fibers. Therefore, it is expected that ultrasonic scattering from myocardium quantitatively characterizes the condition of myocardium. To investigate the angle dependence of ultrasonic scattering in relation to the fiber direction, in this study, ultrasonic echoes from a wire phantom (thinner than a wavelength in diameter) that mimics myocardium were measured as a function of the insonification angle. Two ultrasonic transducers of 7 MHz were employed for transmitting and receiving ultrasonic pulses. Focal points of these transducers were set at the same point on the phantom. A custom-made experimental system can change the azimuth and elevation angles while keeping the focal points at the same point. Measured ultrasonic scattering showed a significant change depending on the azimuth angle. Such changes depending on the azimuth angle were decreased by increasing the elevation angle. These preliminary findings support the hypothesis that the change in the angle between the ultrasonic beam and myocardial fibers due to the heartbeat varies the ultrasonic scattering property.

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Session 5pMU

Musical Acoustics: Simulation and Measurement Techniques for Musical Acoustics II

Shigeru Yoshikawa, Cochair

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Gary P. Scavone, Cochair

McGill Univ., Music Technology, Schulich School of Music, 555 Sherbrooke St., W., Montreal, Quebec H3A 1E3, Canada

Murray D. Campbell, Cochair

Univ. of Edinburgh, School of Physics, Mayfield Rd., Edinburgh EH9 3JZ, U.K.

Invited Papers

1:00

5pMU1. A hybrid approach for simulating clarinetlike systems involving the lattice Boltzmann method and a finite difference scheme. Andrey Ricardo da Silva and Gary Scavone (Computational Acoust. Modeling Lab., McGill Univ., 555 Sherbrooke St. W., Montreal, CA H3A 1E3, Canada, andrey.dasilva@mail.mcgill.ca)

The lattice Boltzmann (LB) method is becoming an increasingly important technique for the simulation of problems involving the interaction between acoustic and fluid fields. New LB algorithms allow the representation of moving boundaries at low Mach numbers. In this work we present the implementation of a hybrid model involving one such LB algorithm coupled to a finite difference scheme to simulate the fluid-structure interaction at the reed of a two-dimensional clarinetlike system. The results of the simulation for different input pressures and reed parameters are compared with the results provided by the related literature. [The first author was supported by CAPES (Brazil).]

1:20

5pMU2. Three-dimensionality and temporal variability of the jet flow in organ pipes. Shigeru Yoshikawa, Yumiko Sakamoto (Dept. of Acoust. Design, Grad. School of Kyushu Univ., Fukuoka 815-8540 Japan), Andreas Bamberger (Albert-Ludwigs-Univ. Freiburg, D-79104 Freiburg, Germany), and Judit Angster (Fraunhofer Inst. for Bldg. Phys., D-70569 Stuttgart, Germany)

Flow visualization by the particle image velocimetry (PIV) is expected to provide quantitative information based on high resolution from the use of sheetlike laser pulses. This PIV technique is applied to investigate the difference in jet-flow structure of organ pipes when the ear (small plate standing at each side of the mouth) is used or not. The experimental results given by PIV are also compared with those by the hot-wire anemometry. Two-dimensionality and laminar-flow condition tend to be maintained when the ear is used. However, when the ear is not applied, the deviation from laminar flow and the resulting three-dimensionality are apparent halfway between the flue and the edge (*re* less than 1800). The ear seems to block the entrainment from the mouth side. Moreover, it is shown that the magnitude of jet velocity is significantly reduced at the phase when the jet deflects innermostly to the pipe with the ear and at the phase when the jet deflects outermostly from the pipe without the ear. These 3D and temporal variability of the jet flow should be examined in more detail in the context of harmonic structure of organ tones and vortex-sound theory on flue instruments.

1:40

5pMU3. Vortex sound and the flute. Andreas Bamberger (Phys. Institut der Universitaet Freieburg, Hermann Herder Str. 3, 79104 Freiburg, Germany)

The flute is investigated for its aeroacoustical properties based on the vortex sound theory. Particle image velocimetry (PIV) is used for quantitative flow determination of the jet-edge interaction. The Endoscopic-PIV offers a nonobstructive view of the system over all phases. The evaluation of the source term through the vorticity is done according to M. Howe (1975). The acoustic flow across the embouchure is determined by excitation through the foot. The flute is operated near 1200 Hz with various jet speeds. Finally the acoustic radiation power in the far field is determined to be compared with the source terms. The following findings are presented: (*i*) The space integrated and time averaged power of the coherent source terms turns out to be positive, i.e., emitting acoustic energy. (*ii*) There is a dominant contribution near the labium. (*iii*) The source term power compares favorably with the far field power, which is well below 1% of the total input power. The results demonstrate that the vorticity of the upper and lower shear layers of the jet cancels to a great extent, except for regions where the transverse amplitude of the jet is large.

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2:00

5pMU4. Sound generation in flutelike instruments: Simulation and analysis. Helmut Kuehnelt (Inst. of Musical Acoust., Univ. of Music and Performing Arts Vienna, Singerstrasse 26a, A-1010 Vienna, Austria)

Three-dimensional flow-acoustic simulations using the lattice Boltzmann method were used to study the sound generation in flutes and recorders. Theory suggests that the fluctuating Coriolis force in the mouth of the instrument caused by the fluctuating vorticity distribution of the jet is the main source of sound. A numerical experiment was undertaken to clarify this hypothesis for flute instruments. In the mouth of the instrument an equivalent temporal fluctuating force was imposed in a quiescent fluid. Although this force excited fluid dynamic effects in the mouth not present at mean flow, it nevertheless induced a standing wave in the instrument with a sound pressure whose amplitude differed only less than 1 dB from the playing situation. This shows the cardinal role of the Coriolis force for the sound generation in flutes. The temporal and spatial structure of the Coriolis force distribution and the vortex sound power generated will be compared for flute and recorder. Especially the spatial structure differs systematically for the two types of instruments. Implications for the design of flute head joints will be discussed.

Contributed Paper

2:20

5pMU5. Particle image velocimetry measurement and vortex sound analysis of high-amplitude acoustically generated flows in open ended tubes. David Skulina (Acoust. and Fluid Dynam. Group, School of Phys., Univ. of Edinburgh, Scotland)

Periodic vortex shedding induced by an acoustic field has been observed by several authors at orifices and at the open ends of tubes. It is a well-known dissipative mechanism and affects the sound field at the open end of a tube and the efficiency with which a standing wave can be maintained within it. This has obvious implications in the playability and design of musical wind instruments. To quantify the losses associated with vortex generation and shedding at the open end of a tube a method for the evaluation of PIV velocity maps using vortex sound theory is outlined. Comparison between the results of the vortex sound calculations and the velocity fields measured from the PIV vector maps allows both qualitative and quantitative descriptions of the acoustical effect of vortex shedding on losses over the acoustic cycle. It is found that the net acoustical losses due to the interaction of vortices with the acoustic field are most significant in a region within 2.5 mm of the open end. Out of this region, vortices act as a net source of sound, but are of lesser magnitude. [Work supported by EPSRC.]

2:35–2:50 Break

Invited Papers

2:50

5pMU6. Synthetic percussion. Stefan Bilbao (Dept. of Music, Rm. 7306B, James Clerk Maxwell Bldg., King's Bldg., Mayfield Rd., Edinburgh EH3 9JZ, Scotland)

The widespread availability of fast small computers has removed many of the restrictions that governed the development of sound synthesis methods in the past, and, in particular, those based on physical modeling of musical instruments. Even relatively complex models can now produce sound output in something approaching real time, if not real time itself. As an example, various models of strings and 2-D percussion instruments, simulated using direct time domain methods, are presented here. Various issues are discussed, including interconnections of various vibrating objects, extensions of percussion instruments involving acoustic "preparation," the modeling of nonlinearities, multichannel output, algorithmic complexity, and operation counts, as well as current work towards developing real time implementations. Sound examples are presented.

3:10

5pMU7. Simulation of piano strings and a soundboard by a large deformation theory. Isoharu Nishiguchi, Masataka Sasaki, and Aki Yamamoto (Kanagawa Inst. of Technol., 1030 Shimoogino, Atsugi, Kanagawa, Japan 243-0292, nishiguc@sd.kanagawa-it.ac.jp)

Three-dimensional finite element analyses of piano strings and a soundboard are presented in which a large deformation theory is adopted. In the large deformation theory, the momentum balance of a body is preserved in the course of the deformation and the changes of direction and amplitude of axial force due to deformation are taken into account. It is shown that the secondary partials appear in the spectrum of the velocity component in the axial direction as well as the transverse direction. A hammer is modeled by masses with springs and dashpots. Gap elements are employed to simulate the contact and detachment between the hammer model and a string. It is demonstrated that multiple reflections between hammer and agraffe can be reproduced by the method. Modeling of the interaction among strings, a bridge, and a soundboard is also discussed in which the crown and the preloading of the soundboard are taken into account.

3:30

5pMU8. Physical modeling of the piano. N. Giordano (Dept. of Phys., Purdue Univ., West Lafayette, IN 47907, giordano@purdue.edu)

As part of an ongoing project we have constructed a computational model of the piano. This model uses Newton's laws to compute the motion of the hammers, strings, soundboard, and room air, yielding the sound pressure that reaches a listener. The model is based heavily on corresponding experimental studies of hammers, strings, and soundboard properties. To date we have focused on the modern piano, but our basic modeling approach can also be used to investigate how changes in the instrument would change the resulting tones. After reviewing the model, we describe studies of the sound produced by models of historical pianos, i.e., the type of piano made in the late 1700s or early 1800s, and compare those tones with the tone of a modern instrument.

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3:50

5pMU9. Numerical dispersion of finite difference method. Yoshimitsu Takasawa (Univ. of Electro-Commun. 1-5-1 Chofugaoka, Chofu, Tokyo, Japan, takasawa@ice.uec.ac.jp)

The partial frequencies of the transverse vibrations of a string, which has stiffness ϵ and is assumed to be hinged at both ends, form an inharmonic series and obey the equation $f_n = f_0 n(1+\pi^2 \epsilon n^2)^{1/2}$, where f_0 is the fundamental frequency. For musical synthesis, the equations of a physical model can be simulated in the time domain using numerical methods. The most straightforward approach can be said to be the finite difference method. The equations are formulated in discrete form using discrete positions $i\Delta x$ and discrete time steps $n\Delta t$. In the finite difference method, two problems must be taken into account. The first is numerical stability: it must be roughly $\gamma > N$. Here, $\gamma = f_e/2f_0$, $f_e = 1/\Delta t$ is the sampling frequency, and $N = L/\Delta x$ is the segment number of the string. The second is the numerical dispersion. It has been shown that some undesirable dispersive effects might be present in the solution if a finite difference scheme is used. This numerical dispersion can be formularized as the equation $f_n \approx f_0 n[1+\pi^2\{\epsilon - \frac{1}{12}(1/N^2) - 1/\gamma^2)\}n^2 - \pi^4/6\{\epsilon/N^2 - \frac{1}{60}(1/N^4 - 1/\gamma^4)\}n^4]^{1/2}, n \leq N-1$.

4:10

5pMU10. Imaging vibrations and flow using electronic speckle pattern interferometry. Thomas R. Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789)

Electronic speckle pattern interferometry has been used to measure sub-micrometer displacements and visualize the deflection shapes of harmonically vibrating objects for many years; however, to date the technique has not been widely used within the community of musical acousticians. The cost of the equipment, the need for effective vibration isolation, and the complexity of the hardware are usually cited as the reasons why this technique has not been more widely utilized in the study of musical instruments. To alleviate these concerns, a speckle pattern interferometer has been designed that is relatively inexpensive, easy to assemble, and very robust. Construction of the interferometer will be discussed, and applications to problems in musical acoustics will be presented.

4:30

5pMU11. Psychoacoustic experiments with virtual violins. Claudia Fritz, Ian Cross (Faculty of Music, Univ. of Cambridge, West Rd., Cambridge, CB3 9DP, UK), Brian C. J. Moore, and Jim Woodhouse (Univ. of Cambridge, Cambridge, CB3 9DP, UK)

This study is the first step in the psychoacoustic exploration of why some violins sound better than others. A method was used that enabled the same performance to be replayed on different ''virtual violins,'' so that the relationships between acoustical characteristics of violins and perceived qualities could be explored. Recordings of real performances were made using a bridge-mounted force transducer, giving an accurate representation of the signal from the violin string. These were then played through filters corresponding to the admittance curves of different violins. Initially, the limits of performance in detecting changes in acoustical characteristics have been characterized. Thresholds were measured for the detection of different modifications of a violins acoustical response, such as a shift in frequency or an increase in amplitude of one or several modes, using a three-interval forced-choice discrimination task. Thresholds were higher for an input of a musical phrase than for a single note, but depended strongly on the choice of the note. The lowest threshold corresponded to a simultaneous shift in frequency of 1.5%, or an increase in level of 3 dB, of several modes, but thresholds appear to be dependent on the musical training of the listeners.

4:50

5pMU12. Use of the energy decay relief (EDR) to estimate partial-overtone decay times in a freely vibrating string. Nelson Lee and Julius O. SmithIII (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), The Knoll, 660 Lomita, Stanford, CA 94305-8180, USA, nalee@stanford.edu)

The energy decay relief (EDR) was proposed by J. M. Jot [IEEE, ICASSP (1992)] for displaying the impulse response of artificial reverberation systems. The EDR is a frequency-dependent generalization of Manfred Schroeder's energy decay curve (EDC), defined at time n as the sum of squared impulse-response samples from time n until decay is complete. The EDR is similarly defined for each band in a uniform filter bank, typically implemented using the short-time Fourier transform (STFT). In this work, we apply the EDR to the problem of estimating decay times for the partials of a freely vibrating string. Previously, such decay times have been measured based on STFT magnitude data. We show that the EDR has certain advantages over the STFT, such as being less sensitive to ''beating'' in the amplitude envelopes of the partial overtones. Results in the context of virtual acoustic guitar modeling will be presented.

Contributed Paper

5:10

5pMU13. Numerical analysis of the 48-key experimental piano. Kenji Hasegawa, Isoharu Nishiguchi, and Masataka Sasaki (Kanagawa Instutute of Technol., 1030 Shimoogino, Atsugi, Kanagawa, 243-0292, Japan, s064817@cce.kanagawa-it.ac.jp)

An experimental piano has been built as a part of research program for students in the 4th grade. Though the main purpose of the building was to use the piano in vibrational and acoustical experiments, it was also intended to deepen the understanding of the structures of a piano by newly designing and building a piano that was different from existing pianos. The range covered by the piano is 4 oct (key nos. 21–68). After the sizes and arrangements of the strings were determined, the metal frame that sustains the tension of the strings was designed. The deformation and the stress were evaluated by the FEM. In this paper, the design and results of numerical analysis of the strings and soundboard are reported.

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Session 5pNSa

Noise and Physical Acoustics: Prediction and Propagation of Outdoor Noise II

D. Keith Wilson, Cochair
U.S. Army Cold Regions Research Lab., Engineering Research and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290

Kohei Yamamoto, Cochair
Kobayashi Inst. of Physical Research, 3-20-41 Higoshi-Motomachi, Kokubunnji-shi, Tokyo 185-0022, Japan

Contributed Papers

1:00

5pNSa1. A predictive noise model for the resurfacing of Nebraska Highway 2. Dominique Cheenne, Connie Lee, Michael Cappiello, Sam Lalk, Coleman Martin, and Philip Muzzy (Dept. of Audio Arts & Acoust., Columbia College Chicago, Chicago, IL 60610)

A detailed investigation of the noise soundscape resulting from traffic on a 3-mile segment of Nebraska Highway 2 prior to resurfacing work was undertaken by the students enrolled in the Acoustics Program at Columbia College, Chicago. Noise data were acquired at six different locations using established protocols and included daytime and nighttime conditions. The test data were compared to the predictions from an outdoor noise propagation computer model and the results were found to be within ± 2 dB for all locations under consideration. The computer model was then used to assess expected noise conditions under various concrete and asphalt scenarios for resurfacing. Data show that reductions of up to 6 dB in the overall noise levels may be achievable depending on the choice for the surface of the roadway.

1:15

5pNSa2. Experimental determination of the correction for noise levels assessed in front of a facade. Gianluca Memoli (Memolix Environ. Consultants, Pisa, Italy), Stylianos Kephalopoulos, Marco Paviotti (Joint Res. Ctr. of the European Commission, Ispra, Italy), and Gaetano Licitra (Tuscany Region Agency for Environ. Protection, Firenze, Italy)

The assessment of noise levels, in the proximity of a building or on its facade, is a requirement of the European Environmental Noise Directive 2002/49/EC concerning environmental noise produced by road and railway traffic, airports, and industries. One of the problems that concerns both theoretical models and measurements is to estimate the correction due to reflections. Several works in the past have approached this problem, now stated by international standards, which suggest using a $+3$ or $+6$-dB correction in function of the microphone position. A complete set of measurements have been performed in the framework of the IMAGINE project for road traffic, in different urban situations, in order to determine the real intereference pattern in terms of coherence length. A careful treatment of the acquired data highlighted a dependence of corrective factors on the distance from the building facade that cannot be neglected. Practical solutions and uncertainties also will be discussed in this work.

1:30

5pNSa3. An exact point-source starting field for the Green's function parabolic equation in outdoor sound propagation. Kenneth Gilbert and Xiao Di (Univ. of Mississippi, Natl. Ctr. for Physical Acoust., Coliseum Dr., Universtiy, MS 38677)

A method for exactly representing a point source in a Green's function parabolic equation calculation is presented. The method is no more difficult to use than standard starting-field formulations and has the several advantages: (1) the point-source solution is essentially exact; (2) only the propagating Fourier components are used; and (3) there is no limitation on the source height, which can be taken to be at $z=0$, if necessary. The method is derived analytically and examples of the numerical implementation are given. [Research supported by U.S. Army Armament, Research Development and Engineering Center (ARDEC).]

1:45

5pNSa4. Predicted effects of forest stand age on acoustic propagation. Michelle Swearingen (Construction Eng. Res. Lab., P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@erdc.usace.army.mil)

A theoretical study on the impact of forest stand age on acoustic propagation will be presented. A Red Pine forest was simulated at 10, 20, 30, 40, and 80 years of age, with assumptions that it was being maintained for utility pole harvest. Forest parameters, such as density, height, and diameter of trees, were used to predict vertical sound-speed profiles and then acoustic propagation. The resulting spectra were weighted to simulate low-frequency (30 Hz) and midfrequency (500 Hz) blast signatures. Spectra and sound-exposure levels (SEL) were examined to determine whether the forest stand age has a significant impact on acoustic propagation within a forest.

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Session 5pNSb

Noise and ASA Committee on Standards: Consumer Product Noise

Matthew A. Nobile, Cochair

IBM Hudson Valley Acoustical Lab., M/S P226, Bldg. 704, Boardman Road Site, 2455 South Rd., Poughkeepsie, NY 12601

Takeshi Toi, Cochair

Chuo Univ., Dept. of Precision Mechanics, Faculty of Science and Engineering, 1-13-27 Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan

Chair's Introduction—2:15

Invited Papers

2:20

5pNSb1. Noise emissions from powered hand tools: A consumer alert!. Charles Hayden and Edward Zechmann (NIOSH, 4676 Columbia Pkwy., C27, Cincinnati, OH 45226)

Existing standards for acquiring sound power levels of powered hand tools specify, for the most part, that sound levels be gathered in the unloaded test condition. However, there is an increase in sound level when the power tool is tested in the loaded condition. There are two purposes for gathering sound power level data: (1) determination of hearing loss hazard or irritation and (2) part of an engineering design optimization effort. The former is the focus of the National Institute for Occupational Safety and Health efforts. Sound power level data were gathered in the loaded and unloaded conditions from a variety of electrically powered hand tools. A model was then used to estimate the A-weighted sound pressure level at the operator's ear from the A-weighted sound power (LWA) measurements of powered hand tools. The magnitude of LWA is found to be a reasonable predictor of the magnitude of sound pressure level exposure that a powered hand tool operator might experience across a variety of acoustical environments. This presentation explores the methods used to gather sound power levels, an evaluation of the model, and an examination of differences found between loaded and unloaded tool test conditions.

2:40

5pNSb2. Protocol for measuring dishwasher sound power levels. Kevin Herreman and Richard Godfrey (Owens Corning Corp., 2790 Columbus Rd., Rte. #16, Granville, OH 43023, kevin.herreman@owenscorning.com)

A leading retailer of home appliances came to the OC acoustics laboratory to evaluate the noise generated by dishwashers sold in their stores. A protocol based on a methodology used in Europe was used. It was modified to account for North American building standards. The standard provides an overall sound power level determination for the complete dishwasher cycle with and without the drying portion of the cycle. As a result customers making a purchasing decision can compare the noise performance of the various products using the same yard stick.

3:00

5pNSb3. Noise exposure limits and measurements for earphones. K. Allen Woo (Plantronics, Inc., 345 Encinal St., Santa Cruz, CA 95060)

There are noise exposure concerns regarding the use of audio earphones and telephone receivers. The permissible noise exposure limits have been defined for the occupational environment for many years, for instance, the US Code of Federal Regulations, 29CFR1910.95, Occupational noise exposure and the Directive 2003/10/EC of European Parliament on Minimum Health and Safety Requirements. Those permissible noise exposure limits are defined for noise exposure in an ''open-field'' environment. An open-field environment is an environment where the noise sources are at a distance from a person's ear. The sound or noise field can be a combination or any of a free field, or partially reflected, diffused, and reverberant fields. Nonetheless, the noise exposures from earphones are not in an open field. They are localized at or inside the user's ear. Conventional open-field noise exposure measurement methods do not apply. During the last 15 years or so, a few different test methods for measuring noise exposure from earphone or telephone receivers were developed. The standards of ITU-T P.360, UL 60950-1, IEEE 269, and EN 50332 were developed mostly by the telephone industry. This presentation addresses the common permissible noise exposure limits, how to relate the limits to earphones, and how to measure them.

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3:20

5pNSb4. Sound quality evaluation of transient sounds in precision information equipments. Masao Yamaguchi (Res. & Development Ctr., Toshiba Tec Corp., 6-78, Minami-cho, Mishima-shi, Shizuoka-ken, 411-8520 Japan), Sho Shirakata, and Takeshi Toi (Chuo Univ., Bunkyo-ku, Tokyo 112-8511, Japan)

These days, sound quality in mechanical products has become a very important factor in addition to reduction of the sound pressure level. A new method is proposed for sound quality evaluation including the transient sound in precision information equipment taking MFP (multifunction peripheral), for example. The MFP has many functional parts and a lot of operational modes, so single copy mode and ten times continuous copies mode in monochrome are taken into account. Since the operational sound including the transient sound is changed with time in one cycle, it is necessary to identify high contribution sound section to decide the sound quality by pair comparison method and SD method with original, filtered, and edited sounds. As a result, sound quality of the transient sound is influenced by the cycle time, the rhythm, the sound pressure, and the frequency characteristics. As the single copy mode, main uncomfortable sound can be estimated at the steady-state sound section caused by a rotational component. On the other hand, for the ten times continuous copies mode, it can be estimated at the particular transient sound section in one cycle caused by collision with mechanical parts, which is different from the single copy mode.

3:40

5pNSb5. Sound source identification for precision information equipment having many kinds of sound sources. Sho Shirakata (Dept. of Precision Mech., Chuo Univ., 1-13-27 Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan), Masao Yamaguchi (Toshiba Tec Corp., Shizuoka-ken, 411-8520, Japan), and Takeshi Toi (Chuo Univ., Tokyo 112-8551, Japan)

These days, precision information equipment having multifunction with high qualities in an office environment is demanded. Sound quality in this equipment has become a very important factor in addition to reduction of the sound-pressure level. MFP (multifunction peripheral) has many functional parts and operational modes, so it makes uncomfortable transient sounds in one cycle. The primary sound section to decide the sound quality of the MFP was estimated by sound quality evaluation as in previous research. Main uncomfortable sounds could be confirmed at the steady-state sound section as the single copy mode and at the particular transient sound section in one cycle as the ten-times continuous copies mode. So, it is necessary to identify the position of uncomfortable sound sources at each mode for comfortable sound design. Principal component analysis (PCA) using acceleration measured by accelerometers and sound pressure measured by microphones is applied to identify the number of sound sources. Sound-pressure distribution in one cycle from the MFP having many kinds of sound sources can be measured by microphone array at each moment. As a result, the sound source position can be identified by using PCA and sound-pressure distribution.

4:00

5pNSb6. The role of noise audits in characterizing and reducing consumer product noise. David L. Bowen (Acentech Inc., 33 Moulton St., Cambridge, MA 02138)

Consumer products often share some common traits that impact both the measurement techniques that can be employed, as well as the noise control options that are available. These traits include their relatively small size (except for the special case of automobiles), low profit margin, and quick time to market. Their small size implies that existing units or prototypes can usually be set up for acoustical measurements inside a modestly sized reverberation room. The low profit margin implies that design solutions, rather than adding "acoustical" materials, are the preferred route to a quieter product, but the quick time to market can sometimes work against this approach. This talk will focus on a sound power based "noise audit" technique that is useful for characterizing and rank ordering the potential noise sources in a product, in order to devise the most effective noise reduction strategies. Case studies will be used to illustrate the use of this engineering technique applied to the particular case of consumer products, where the design trend is often towards lighter, stiffer components and faster rotating parts, all of which tend to increase noise.

4:20

5pNSb7. Consumers, products, and noise: The economic, social, and political barriers to reducing noise in consumer products sold in North America. Leslie Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601)

The paper examines the economic, social, and political barriers to reducing noise in consumer products sold in North America. Included in the economic issues are the cost of noise control and whether or not consumers are willing to pay more for a quieter product. An equally important economic barrier, however, is that the marketplace currently does not have an effective mechanism to reward builders of quieter products, even if noise treatments were cost effective. There is very little information available to consumers about the noise levels of consumer products. Without their informed decisions in the marketplace, no signal reaches the manufacturers. Complicating this problem is the consolidation of the consumer product manufacturing sector and the consolidation in the retail sector. The social barriers include the public's attitude towards noise (that it is an unavoidable cost of living in a technologically advance society) and the failure to see noise as a pollutant (that it is merely something unwanted). Political barriers include an absence of government involvement in consumer product noise, in terms of regulation, labeling, as well as a general concern for the issue.

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Contributed Paper

4:40

5pNSb8. Evaluation of noise intensity level using an eccentric press during the production of metallic closure: Findings according to different positions surrounding the press. Paulo H. T. Zannin (Lab. de Acustica Ambiental, Industrial e Conforto Acustico, Dept. Eng. Mecanica, Univ. Federal do Parana, C. Politecnico, Curitiba, Pr, Brazil, 81531-990) and Samuel S. Ansay (Univ. Federal Tecnologica do Parana, Centro, Curitiba, Pr, Brazil, 80230-901)

This study was performed to evaluate the differences in noise level produced by an eccentric press, during the manufacturing of metallic closures for steel cans. The primary objective of the study was to establish a rank of sound-intensity level and sound-power level as well as the global noise level emitted by this particular press. Ten different areas in the surface surrounding the press were evaluated regarding both the sound power and the sound intensity of each measurement. Each site was measured twice to increase accuracy. The ten areas sampled were sweeping using intensity probe. We have found the surface area to demonstrate the lowest noise level was at S1 and the highest one was at S10 (p value 0.0005). In conclusion, our study demonstrates adequately the global noise level as well as in specific sites, indicating areas requiring special treatment for noise reduction in this environment.

SATURDAY AFTERNOON, 2 DECEMBER 2006 WAIANAE ROOM, 1:15 TO 5:15 P.M.

Session 5pPA

Physical Acoustics: Land Mines and Ultrasonics

Masahiro Ohno, Cochair
Chiba Inst. of Technology, 2-17-1 Tsudanuma, Narashino, Chiba 275-0016, Japan

Roger M. Waxler, Cochair
Univ. of Mississippi, National Ctr. for Physical Acoustics, 1 Coliseum Dr., University, MS 38677

Contributed Papers

1:15

5pPA1. Optimizing seismo-acoustic sonar source array configuration for maximum radiated Rayleigh wave energy. Anthony N. Johnson (Dept. of Appl. Mathematics, Naval Postgraduate School, Monterey, CA 93943), Clyde L. Scandrett, and Steven R. Baker (Naval Postgraduate School, Monterey, CA 93943)

A three-dimensional (3-D) continuum mechanics approach to the development of a time-dependent finite element model for optimizing the position and excitation of a seismo-acoustic sonar source array to detect the presence of buried landmines will be presented. Various source configurations will demonstrate the use of constructive and destructive interference, which maximizes the radiated energy of unidirectional Rayleigh waves while suppressing the radiation of body waves. Radiation characteristics are analyzed in a linear, horizontally stratified (isotropic and homogeneous within each layer) half-space with a discrete number of transient seismic sources. Results for Rayleigh wave strengths are presented in both a homogeneous half-space and a layered medium.

1:30

5pPA2. Nonlinear acoustic experiments involving landmine detection: Status and future. Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD 21402) and James M. Sabatier (Univ. of Mississippi, University, MS 38677)

In past experiments [J. Acoust. Soc. Am. **116**, 3354–3369 (2004)], airborne sound at two primary tones, f_1 and f_2 (closely spaced near resonance), undergo acoustic-to-seismic coupling. Due to interactions with the buried plastic landmine, a scattered surface profile can be measured. Particle velocity profiles at f_1, f_2, and $f_1-(f_2-f_1)$ exhibit single peaks; those at $2f_1-(f_2-f_1)$, f_1+f_2, and $2f_2+(f_2-f_1)$ involve higher-order mode shapes. For some combination frequencies the "on target" to "off target" contrast ratio can improve by over 20 dB. Near resonance, the bending (softening) of a family of increasing amplitude tuning curves, involving the vibration over the landmine, exhibits a linear relationship between the peak particle velocity and corresponding frequency. Hysteresis effects and slow dynamic behavior have also been observed. The interaction between the top-plate interface of a buried landmine and the soil above it appears to exhibit many characteristics of the mesoscopic/nanoscale nonlinear effects that are observed in geomaterials like rocks (sandstone). In linear detection schemes the natural inhomogeneous soil layering can generate some false alarms. Recent "off target" versus. "on target" tuning curve measurements show a substantial increase in nonlinearity, suggesting this to be a good false alarm discriminator. [Work supported by ONR.]

1:45

5pPA3. Phase inversion and two-frequency interaction in nonlinear time-reversal acoustic method of land mine detection. Alexander Sutin (Artann Labs., Trenton / Stevens Inst. of Technol., Hoboken, NJ), Brad Libbey (U.S. Army RDECOM CERDEC NVESD, Fort Belvoir, VA 22060), and Armen Sarvazyan (Artann Labs., Inc., West Trenton, NJ 08618)

Seismic waves are known to generate a nonlinear response in soil, but data show that the presence of a mine increases the nonlinearities due to interactions of the mine and soil system. The nonlinear effects were investigated by two methods. The first, phase inversion, uses two short, broadband TRA focused signals with opposite signs. These are broadcast sequentially and the responses are added in post processing. This results in cancellation of the linear response leaving the nonlinear system components. The second method relies on two-frequency intermodulation. In this case time-reversed seismic energy at 280 and 350 Hz was used to excite the system while the response was analyzed at the sum frequency, 630 Hz. The experiments demonstrate that the presence of the mine increases nonlinearity by 12 dB for the phase-inversion method and by 6 dB for two-frequency interactions in comparison with intact sand. These results support the hypothesis that the interface between mine body and surrounding sand significantly increases the nonlinear response as compared to sand alone. Work was supported by U.S. Army RDECOM CERDEC Night Vision and Electronic Sensors Directorate.

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2:00

5pPA4. Time-reversal acoustic focusing of waves produced by seismic vibrators. Alexander Sutin, Victor Kurtenoks (Artann Labs., Trenton/Stevens Inst. of Technol., Hoboken, NJ), James Sabatier, Rick Burgett, Mike ONeil, Slava Aranchuk (Univ. of MS, University, MS 38677), Brad Libbey (U.S. Army RDECOM CERDEC Night Vision and Electron. Sensors Directorate, Fort Belvoir, VA), and Armen Sarvazyan (ARTANN Labs., Inc., NJ 08618)

A land mine detection system has been developed based on excitation of seismic waves by four mechanical shakers and measurements of ground vibration by a 16-beam, scanning laser Doppler vibrometer (LDV). Time-reversal acoustic (TRA) focusing was applied to concentrate seismic waves using the above system. Linear swept frequency signals and orthogonal initial signals in the frequency band of 100–500 Hz were used in outdoor testing. The system impulse response from any shaker to the LDV output was computed by cross correlating the original and recorded signals. With orthogonal signals, the system impulse responses between multiple shakers and the LDV were measured simultaneously, thus reducing the overall measurement time. Each shaker then reradiates the time-reversed impulse responses focusing the seismic energy to a point on the ground. Experiments demonstrate that the TRA focusing provided a high concentration of elastic wave energy in the test area with typical focusing spots about 20 cm in diameter. [Work supported by U.S. Army RDECOM CERDEC Night Vision and Electronic Sensors Directorate.]

2:15

5pPA5. Acousto-optic interactions at a surface perturbed by acoustic and gravity waves. James Churnside, Hector Bravo (ESRL/Zeltech, 325 Broadway, Boulder, CO), Konstantin Naugolnykh, and Joseph Fuks (Univ. of Colorado, Boulder, CO)

We studied the acousto-optic interaction at a water surface perturbed by an underwater sound source. Using a coherent optical detection technique, we measured the power spectra of the scattered light. Filtering of the power spectra allowed us to detect the sound-induced perturbations of the surface in the presence of surface waves, even though the amplitude of the waves was much larger than the acoustic perturbations of the surface. The theory of scattering by a surface perturbed by sound and by surface waves is developed and that theory is compared with the experimental results.

2:30

5pPA6. Improvement of dispersion of nanometer-sized diamond particles for precise polishing by ultrasound exposure. Takeyoshi Uchida, Tsuneo Kikuchi (Natl. Metrology Inst. of Japan, Natl. Inst. of Adv. Industrial Sci. and Technol., Japan), Yoshikazu Setoyama, Norimichi Kawashima, and Shinichi Takeuchi (Toin Univ. of Yokohama)

A novel acoustic dispersion method for fine diamond particles was investigated by our group. Nanometer-sized diamond particles for abrasive agents should be dispersed sufficiently in water without aggregation. The particles usually aggregate immediately after their manufacture. The dispersants and dispersion equipment are used for dispersion of the particles. However, these dispersion methods present problems in their dispersion characteristics and reaggregation because such methods do not improve the essential surface characteristics. We suggested the acoustic dispersion method. We reported in 2005 at the IEEE Conference in Rotterdam that nanometer-sized diamond particles with 5-nm primary particle size were dispersed using ultrasound. The particles with 5-nm primary particle size are anticipated for use as future abrasive agents. Particles with 50-nm primary particle size are actually used at present as abrasive agents. Therefore, in this study, we investigated the dispersion of particles with primary particle sizes of 20, 50, 80, and 150 nm. Results show that the aggregated particles were disaggregated to nanometer-sized particles. Zeta potentials of the diamond particles were increased by ultrasound. These results suggest the improvement of dispersion of nanometer-sized diamond particles in water.

2:45

5pPA7. Observation of living samples in a C-mode configuration with an ultrasonic phase conjugator. Tatsuya Seki and Masahiro Ohno (Chiba Inst. of Technol., 2-17-1 Tsudanuma, Narashino, Chiba 275-0016, Japan)

Ultrasonic images of living samples have been obtained in a C-mode configuration with phase conjugation (time-reversal) processes. A phase conjugator for 10-MHz ultrasound was realized by the field-mixing in a PZT block pumped with 20-MHz electric fields. Samples, relatively thick (mm to cm) living materials, were placed between an ultrasonic transmitting/receiving transducer and the phase conjugator and were scanned to form C-mode transmission images. Ultrasonic tone-burst pulses radiated onto the sample were led into the PZT block and were converted to their phase conjugate waves, which, for their time-reversing nature, acted to cancel the wavefront deformation introduced by the thick living samples. In the experiment, images were obtained at a frequency of 10 MHz for several samples such as chicken liver, poke meat, and some model samples made of agarose gel, both by the phase conjugate method and by the conventional one. Images by the phase conjugate method showed relatively good accordance with the real ultrasonic attenuation distribution, while those by the conventional method were very sensitive to the shapes or the discontinuities of samples.

3:00

5pPA8. Propagation and interaction of phase conjugate ultrasound waves in moving media. V. Preobrazhensky, P. Pernod, and Yu Pyl'nov (Joint European Lab. LEMAC: Institut d'Electronique et de Micro-electronique et de Nanotechnologie IEMN-DOAE-UMR CNRS 8520, Ecole Centrale de Lille, BP48, 59651 Villeneuve d'Acsq, Cedex, France, Wave Res. Ctr. of General Phys. Inst. RAS and Moscow State Inst. of Radio Eng., Electron. Automation, Moscow, Russia, vladimir.preobrajenski@iemn.univ-lille1.fr)

Wave phase conjugation is known as an effective tool for compensation of phase distortions arising during wave propagation in inhomogeneous media. Backpropagation of the phase conjugate wave (PCW) towards the emitter restores the phase of the primary wave on its source. This feature of PCW results from reversibility of wave processes in stationary media. The presence of flows or of moving objects in a medium breaks the time-reversal invariance of the wave process and leads, in particular, to the phase shift of the emitted and conjugate waves on the source. In the present paper the theory of PCW propagation in nonlinear and moving inhomogeneous media is developed on the basis of the modified KZK approximation. Applications of the wave phase conjugation for new methods of ultrasonic velocimetry and nonlinear diagnostics of flows are considered.

3:15–3:30 Break

3:30

5pPA9. Ultrasonic scattering in polycrystalline material with elongated grains. Goutam Ghoshal and Joesph A. Turner (Dept. of Eng. Mech., W317.4 Nebraska Hall, Univ. of Nebraska-Lincoln, Lincoln, NE 68588-0526)

The scattering of elastic waves in polycrystalline materials is relevant for ultrasonic materials characterization and nondestructive evaluation (NDE). Ultrasonic attenuation and diffuse backscatter measurements have been especially useful for extracting microstructural information such as grain size and for detecting flaws in materials. Accurate interpretation of experimental data requires robust scattering models. Such models often assume constant density and uniform grain size such that the scattering for grains with mainly spherical geometry is well understood. However, many structural materials are processed in such a way that the microstructure has grain elongation (e.g., from rolling). The scattering of elastic waves in such media must be understood well for design of effective materials inspection methods and for quantitative data analysis. For materials with grain elongation, the appropriate spatial correlation function is no longer

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isotropic such that closed-form solutions for attenuation cannot be achieved. Here, attenuation and diffuse backscatter expressions are derived as a function of wave propagation direction with respect to the axis of grain elongation. Numerical results are presented for materials of common interest under assumptions of statistical homogeneity. These results are anticipated to impact ultrasonic nondestructive evaluation of polycrystalline media. [Work supported by U.S. DOE.]

3:45

5pPA10. Accurate time domain computation of linear wave propagation using Chebyshev collocation and matrix diagonalization. Lixi Huang (Dept of Mech. Eng., The Univ. of Hong Kong, Pokfulam Rd., Hong Kong)

Accurate time domain computation of the wave propagation phenomena is of great interest to many fields of studies. The spectral method of Chebyshev collocation is adopted for smooth problems of linear waves. Space differentiation is discretized by Chebyshev derivative matrices and the wave equation is cast as a second-order ordinary differential equation in time domain. The general boundary conditions involving pressure and acoustic particle velocity are discretized and absorbed into the system equations for the unknown vector containing all grid points. The system equations are solved accurately by the method of matrix diagonalization, which involves the finding of the eigenvalues and eigenvectors for the appropriate combinations of the damping and stiffness matrices. For this solution, the time step for stable and accurate computation is only limited by the consideration of round-off errors when the exponential functions are evaluated. Comparison is made with analytical solutions in the one-dimensional case, and good agreement is obtained. The only perceived drawback would be the size of matrices that can be alleviated by the use of domain decomposition approaches. The physical meaning of the eigenvalues derived from the discretization procedure is discussed.

4:00

5pPA11. Ultrasonic characterization of microstructure evolution during processing. Liyong Yang and Joseph A. Turner (Dept. of Eng. Mech., Univ. of Nebraska-Lincoln, Lincoln, NE 68588, jaturner@unl.edu)

Many cold-working processes for polycrystalline metals cause alignment of the grains with a single symmetry axis. This type of microstructure is called fiber texture. The existence of a preferred orientation of the grains has a significant influence on the propagation and scattering of ultrasonic waves, which are often used for material inspection. Knowledge of the wave attenuation of such textured materials is of both theoretical and practical interest to nondestructive testing and materials characterization. In this article, the quantitative relations between fiber texture and wave attenuations of hexagonal crystals are presented. The texture is characterized by a Gaussian distribution function that contains a single parameter that governs the transition of the texture from perfectly aligned crystals to statistically isotropic. Under this assumption, the materials of interest have a varying degree of transverse isotropy representative of processing conditions. Simple expressions for the attenuations of the three modes of waves are given in a concise, generalized representation. Finally, numerical results are presented and discussed in terms of the directional, frequency, and texture dependence. The results presented are expected to improve the understanding of the microstructure evolution during thermomechanical processing. [Work supported by U.S. DOE.]

4:15

5pPA12. Shaped sound focusing using multiple sources: An acoustical ring. Jin Young Park and Yang-Hann Kim (Ctr. for Noise and Vib. Control at KAIST, Japan, jypark1979@kaist.ac.kr)

Shaped sound focusing is defined as the generation of acoustically bright shape in space using multiple sources. The acoustically bright zone is a spatially focused region with relatively high acoustic potential energy level. In view of the energy transfer, acoustical focusing is essential because acoustic energy is very small to use other types of energy. In clinical uses, for example, there are several approaches for using thermal energy from focused ultrasonic wave energy, so-called high intensity focused ultrasound, concerning point-focusing not regional focusing. If sound shape can be controlled, it offers various kinds of solutions for clinical uses and others because a regional focusing is essentially needed to reduce a treatment time and enhance the performance of transducers. For making the shaped-sound field, control variables, such as a number of sources, excitation frequency, source positioning, etc., should be considered according to geometrical sound shape. To verify these relations between them, a wave-number domain matching method is suggested because the wave-number spectrum can provide the information of the source from the phase and magnitude of the wave-number spectrum. In this presentation, discussions about shaped sound focusing and how to make this sound field are provided for the case of an acoustical ring.

4:30

5pPA13. Noncontact acoustic manipulation using a standing wave field in air. Teruyuki Kozuka, Kyuichi Yasui, Toru Tuziuti, Atsuya Towata, and Yasuo Iida (Natl. Inst. of Adv. Industrial Sci. and Technol. (AIST), Shimoshidami, Moriyan-ku, Nagoya 463-8560, Japan, kozuka-t@aist.go.jp)

A noncontact manipulation technique is necessary to develop micromachine technology, biotechnology, and new materials processing. The authors have developed an acoustic manipulation technique to transport particles in water [T. Kozuka, Proceedings of WCU2003, Paris, 483–486 (2003)]. Although it is more difficult to generate a strong sound field in air than in a liquid, many researchers are actively studying the trapping of particles, droplets, and aerosols in air using ultrasound. Nevertheless, many of the resultant studies only trap and observe objects; they do not transport them. This paper describes an advanced manipulation technique to transport small objects in air using that scheme. A standing-wave field was generated by two sound beams (40 kHz) using bolted Langevin transducers. The beams' axes cross each other. Expanded polystyrene chips were trapped at nodes of the sound pressure in the crossing sound beams. The trapped position was shifted by changing the phase difference of the two sound beams. In addition, transportation at constant speed of the trapped target was realized by a slight difference in ultrasonic frequency between the two sound beams. This system is also applicable to liquid droplets and aerosols in air.

4:45

5pPA14. Propagation phase and zeros in the complex frequency plane. Yoshinori Takahashi (Kogakuin Univ., Tokyo, Japan, takahashi@acoust.rise.waseda.ac.jp), Mikio Tohyama (Waseda Univ., Tokyo, Japan), and Kazunori Miyoshi (Kogakuin Univ., Tokyo, Japan)

This paper describes the transfer function (TF) including the coherent sound field as the distance between source and observation points increases, from a point of view of the number of zeros on a complex frequency plane. The propagation phase could be estimated according to the linear-regression analysis for narrow frequency-band frequency characteristics of the minimum-phase component [Inst. Electron. Inf. Commun. Eng. Jpn., **J89-A**(4) 291–297 (2006) (in Japanese with English figures)]. This paper analyzes zeros of the TFs for the coherent and reverberant fields based on the modal and random sound field theory. Consequently the distribution of zeros could be estimated in the transition from the coherent to the reverberant conditions.

5:00

5pPA15. Using instabilities in the long wavelength transverse optical mode to generate nuclear fusion. Brian Naranjo and Seth J. Putterman (Phys. Dept., UCLA, 1-129 Knudsen Hall, Los Angeles, CA 90095)

Various crystals such as lithium niobate or lithium tantalate display a phase transition due to an instability in the long wavelength transverse optical mode. These low-temperature phases develop huge electric fields which can accelerate ions to energies where nuclear fusion is observed [Nature **434**, 1115–1117 (2005)]. [Research funded by DARPA, ONR, NSF.]

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Session 5pSC

Speech Communication: Production II (Poster Session)

Edward S. Flemming, Cochair
Massachusetts Inst. of Technology, Dept. of Linguistics and Philosophy, 77 Massachusetts Ave., Cambridge, MA 02139-4307

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Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

5pSC1. Finite-element method analysis of acoustic characteristics of the vocal tract with the nasal cavity during phonation of Japanese /a/. Hiroki Matsuzaki and Kunitoshi Motoki (Faculty of Eng., Hokkai-Gakuken Univ., S-26 W-11, Chuo-ku, Sapporo 064-0926, Japan)

For this study, the transfer functions and active sound intensities of a vocal tract model with and without a nasal cavity were computed using a three-dimensional finite-element method (FEM). The models were based on vowel data obtained by magnetic resonance imaging (MRI) of the vocal tract with a nasal cavity during phonation of the Japanese /a/. The oral cavity was also coupled with the nasal cavity in a three-dimensional volume of radiation. Effects of wall impedance were also examined. Coupling of the nasal cavity to the oral cavity indicated the following aspects. Additional peaks appeared below 3 kHz for the lossless condition. However, they disappeared in the simulation for the soft wall condition. The sound energy circulation did not occur in the simulation for the soft wall condition. Regarding effects of the wall boundary condition on the spectral envelope, three-dimensional simulations confirmed the upward shift of lower formant frequencies. However, disagreement of the formant frequencies between simulated and real speech should be investigated further by adjusting the wall boundary condition to a more realistic one.

5pSC2. Using tagged cine-MRI and finite-element method to lower bound the number of independently controllable motor units in the tongue. Caroline Essex-Torcaso, William S. Levine (Dept. of Elec. and Comput. Eng., Univ. of Maryland), Maureen Stone, Emi Z. Murano (Univ. of Maryland School of Dentistry, Baltimore, MD 21201), Vijay Parthasarathy, and Jerry L. Prince (Johns Hopkins Univ.)

The biomechanical structure of the tongue is unusual in that there is no rigid structure, such as a bone, for the muscles to act against. In order to understand these biomechanics and their control, a mathematical model has been created. This model has the form of a nonlinear controllable incompressible elastic structure that undergoes large deformations. The model was used to solve a simplified two-dimensional inverse problem. For a simple speech motion (/i-u/), the trajectories of specific points within the tongue were tracked by means of tagged cine MRI. These points correspond to finite element nodes and were used as input to the simplified model. The remaining unknowns in the model are the set of muscle activations that produce the observed motion. Because there is no such set of activations, the set that best approximates the observed motion in a least squares sense was chosen. The results indicate that there must be independently controllable compartments in SL and GG in order to produce this movement. Three methods were used to improve the representation of the tongue and the resolution of the model: refinement of the number of elements, refinement of the number of muscle activations, and the inclusion of dynamics.

5pSC3. Simulation of phonation control using a tensile vocal-fold model. Chao Tao, Yu Zhang, and Jack J. Jiang (Dept. of Surgery, Div. of Otolaryngol. Head and Neck Surgery, Univ. of Wisconsin Med. School, Madison, WI 53792-7375)

The vocal folds act as an energy transducer that converts aerodynamic power into acoustic power. Therefore, modeling the vocal-fold vibration could provide valuable information for voice synthesis. In previous vocal-fold lumped models, the vocal-fold was simplified as a spring oscillator; the restoring force due to lateral deflection of vocal-fold tissue is represented by equivalent lateral springs. However, the vocal-fold tissue actually contains an intricate extracellular matrix, composed of longitudinal elastin, which lies parallel to the vocal fold. Thyroarytenoid, lateral cricoarytenoid, and cricothyroid muscles control the elongation of these longitudinal elastin and further determine the voice pitch, dysphonia, and rough voice. In this study a tensile vocal-fold model is proposed to simulate vocal-fold vibration. In this model the lateral tissue deflection is represented by lateral spring and the longitudinal elastin elongation is described by nonlinear longitudinal springs. Differential equations are developed to describe the tensile vocal-fold model. The fundamental frequency and phonation threshold pressure are predicted by the proposed vocal-fold model. Numerical simulations prove that the proposed tensile vocal-fold model can accurately describe the muscles control on phonation pitch and so on. [Work supported by NIH.]

5pSC4. A new aerodynamic critical parameter of the speech production: Phonation threshold flow. Jack J. Jiang and Chao Tao (Dept. of Surgery, Div. of Otolaryngol. Head and Neck Surgery, Univ. of Wisconsin Med. School, Madison, WI 53792-7375, jiang@surgery.wisc.edu)

In this paper, the implications of phonation threshold flow (PTF) as an addition to the aerodynamic parameters of speech production system are studied. PTF, the minimum airflow volume velocity able to sustain stable vocal-fold vibration, could have utility in clinical vocal disease assessment just like phonation threshold pressure (PTP) [I. R. Titze, J. Acoust. Soc. Am. **83**, 1536–1552 (1988)]. Furthermore, because glottal airflow can be more easily measured noninvasively than subglottal pressure, PTF could be more suitable for routine clinical assessment than PTP. Theoretical

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studies indicate that PTF is a sensitive aerodynamic parameter dependent on tissue properties, glottal configuration, and vocal-tract loading. Accordingly, this theory suggests that PTF is reduced by the following: decreasing tissue viscosity, decreasing mucosal wave velocity, increasing vocal-fold thickness, or decreasing prephonatory glottal area. A divergent glottis and low vocal-tract resistance also reduce phonation threshold flow. Finally, the significance of PTF for investigation of the distribution of energy in a vocal-fold vibration system and the potential for application of the PTF parameter are assessed. [Work supported by NIH.]

5pSC5. Lung pressure dependence of glottal sound source. Hideyuki Nomura and Tetsuo Funada (Div. of Elec. Eng. and Comput. Sci., Kanazawa Univ., Kakuma-machi, Kanazawa-shi 920-1192, Japan, nomu@t.kanazawa-u.ac.jp)

The present study investigates the lung pressure dependence of the vibration of vocal cords using numerical experiments based on our proposed glottal sound source model. The glottal sound source model is described as a coupled problem between unsteady glottal jets and mechanical vocal cords. The vocal cord is assumed to be an elastic cover with effective mass of vocal cord. To simulate the mechanical properties of vocal cords, the elastic cover is supported by distributed small mechanical elements of a spring and damper. The speech production process can be predicted by alternately solving the motion of glottal jets and the vocal cords' vibration. Results of this simulation show that the fundamental frequency of vocal cords' vibration and the propagation velocity of the mucosal wave first increase and then remain constant with lung pressure. The threshold lung pressure of 200–400 Pa and the propagation velocity of mucosal waves is of the order of 1 m/s, which are consistent with measured values. These results suggest that our proposed model is suitable for description of a glottal sound source.

5pSC6. Direct computational method of including piriform fossae and nasal cavity in a time-domain acoustic model of the vocal tract. Parham Mokhtari (NICT, ATR Cognit. Information Sci. Labs., 2-2 Hikaridai Seikacho, Kyoto 619-0288 Japan, parham@atr.jp)

Frequency-domain simulations of the human vocal tract (VT) have previously shown the importance of including the piriform fossae, which impart a pole and two zeros in the 4–5-kHz frequency range and thereby contribute to speaker individualities. The literature has also shown that time-domain simulation of VT acoustics can result in high-quality synthesis naturally including interactions between the time-varying glottal area and the supraglottal VT. In the present work, the time-domain model of [S.Maeda, Speech Commun. **1**, 199–229 (1982)] was extended to include both left and right piriform fossae as side-branches connected to the main VT, in addition to the nasal tract and sinuses. Departing from Maeda's original implementation owing to the complexity of including more than one side branch, the variables representing acoustic pressure and volume velocity at the piriform fossae and nasal tract junctions were analytically eliminated, and the resulting large system of linear equations were solved simultaneously at each simulation sample. This direct method runs at only a few times real-time on a 1.8-GHz notebook PC, while achieving a more natural sound quality in speech synthesis and control over timbral (or voice quality) features that contribute to each speaker's individuality. [Work supported by NiCT and SCOPE-R of Japan.]

5pSC7. Measures of intraoral pressure pulse shape during stop consonants. Jorge C. Lucero (Dept. of Mathematics, Univ. Brasilia, Brasilia DF 70910-900, Brazil, lucero@unb.br) and Laura L. Koenig (Haskins Labs, New Haven, CT)

To maintain voicing during a stop consonant, adult speakers typically perform articulatory actions that increase supraglottal volumes and slow the rate at which of intraoral pressure (Pio) rises. In an influential study, Müller and Brown [in *Speech and Language: Advances in Basic Research and Practice* 4, edited by N. Lass (Academic, New York, 1980), pp. 318–389] proposed a quantitative measure of Pio pulse shape, based on the difference between (a) the slope from pressure at closure to average pressure and (b) from average pressure to pressure at release. We recently applied these techniques to data from women and child speakers of English and observed some shortcomings in Müller and Brown's measurement technique, such that the slope difference angle did not clearly reflect the actual pressure pulse shape. The purpose of this paper is to explore new methods of analyzing intraoral pressure contours, based on fitting a mathematical model to the pressure pulse shapes. It will report our progress and compare our results with the previous technique.

5pSC8. Estimation of area function for three-dimensional vocal tract using finite-element method. Keita Mochizuki and Takayoshi Nakai (The Grad. School of Electron. Sci. and Technol., Shizuoka Univ., 3-5-1, Johoku, Hamamatsu-shi, Shizuoka, 432-8561 Japan)

Numerous studies have obtained a vocal tract area function using data obtained from magnetic resonance imaging (MRI). However, this approach often results in increased error between formant frequencies derived by recorded voice and estimated frequencies derived by MRI. Two factors engender error: creating the vocal tract model and estimating the area function from the vocal tract model. Generally, the errors from those two factors are not considered independently. This study is intended to establish precise estimation of the area function automatically from a vocal tract model. Pressure contours and sound intensity in the vocal tract are obtained using the finite-element method (FEM); the vocal tract area function is obtained from them. The precision is assessed by comparing the formant frequencies that are derived using the FEM with our method. Results show that the vocal tract area function can be estimated precisely from three-dimensional MRI.

5pSC9. Linear stability analysis of an aeroelastic model of phonation. Zhaoyan Zhang, Juergen Neubauer, and David Berry (UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095)

This study presents a linear stability analysis of a two-dimensional aeroelastic model of phonation, which incorporates the interaction between the glottal flow and the vocal folds. The model consists of a wall constriction (shaped by the medial surface of vocal the folds), which is situated in a rigid pipe with an applied potential flow. The vocal-fold constriction is modeled as a plane-strain linear elastic layer. An eigenvalue problem is formulated and the eigenvalues of the coupled system and the eigenshape of the vocal fold surface are calculated. Both linear divergence and flutter instabilities are possible. Phonation onset frequency and pressure are investigated as a function of glottal channel width, vocal fold geometry, and vocal fold material properties. Based on aeroelasticity theory, this model allows the prediction of the effects of changing geometric and viscoelastic properties of the vocal folds on phonation onset.

5pSC10. Comparison of vocal tract shaping patterns derived from articulatory fleshpoint data and MRI-based area functions. Brad H. Story and Kate Bunton (Dept. of Speech, Lang., and Hearing Sci., Speech Acoust. Lab., Univ. of Arizona, Tucson, AZ 85721)

A recent addition to the University of Wisconsin x-ray microbeam (XRMB) database [Westbury, UW-Madison, 1994] (XRMB) is a speaker from whom MRI-based vocal tract area functions have been previously obtained. The purpose of this study was to compare principal components derived from collections of vocal tract shapes for vowels obtained with these two different techniques. For the XRMB data, midsagittal cross-distance functions representative of each of 11 vowels were first obtained from tongue and jaw pellet positions and the hard palate profile of the speaker. A principal components analysis was then performed on the set of cross-distance functions. Results indicated that the first two orthogonal components (referred to as modes) accounted for more than 90% of the

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variance. The shape of these two modes was shown to be similar to those previously determined from the same speakers set of MRI-based area functions. The XRMB-based modes were then used to develop a database-lookup technique where mode coefficients could be determined for time frames throughout a series of vowel sequences. Finally, a procedure is described for using the time-dependent mode coefficients obtained from the XRMB data as input for an area function model of the vocal tract. [Work supported by NIH R01-DC04789.]

5pSC11. Temporal structures of articulatory movements: A contrastive study of stop consonants in Japanese, Korean, and Chinese. Tokihiko Kaburagi and Yoshitaka Nakajima (Faculty of Design, Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540 Japan, kabu@design.kyushu-u.ac.jp)

Stop consonants of Japanese, Korean, and Chinese were studied from an articulatory perspective, particularly in relation to the temporal structure of articulatory movements. In general, acoustical features of stop consonants, such as the amount of aspiration, voice onset time (VOT), and so on, are largely controlled by interarticulatory movements of the tract and glottis. In this study, several speakers were instructed to produce stop consonants of their native tongues. Then, states of the articulators were monitored using an electroglottograph (EGG) and pressure and flow sensors. While the EGG tracked vibratory patterns of the glottis, the pressure and flow sensors were used to measure the air pressure inside the vocal tract and the amount of air flowing out of the mouth. It is noteworthy that these languages have different sets of stops: the voiced and unvoiced stops of Japanese; the aspirated, weak, and strong stops of Korean; and the aspirated and nonaspirated stops of Chinese. Temporal structures of interarticulatory movements were analyzed from the observed data. Their respective acoustical consequences were compared among these languages. [Work supported by JSPS and 21st Century COE, Kyushu University.]

5pSC12. The relation of the temporal variation of F2 to articulator movement. Benjamin V. Tucker (Dept. of Linguist., Univ. of Arizona, P.O. Box 210028, Tucson, AZ 85721-0028, bvt@u.arizona.edu) and Brad H. Story (Univ. of Arizona, Tucson, AZ 85721)

The second formant frequency (F2) has been shown to serve a prominent role in speech perception. The specific relation of the temporal variation of F2 to tongue and lip motion is, however, less well understood. Using the x-ray microbeam database (Westbury *et al.*, 1994), data from six males and six females were extracted for six vowel to vowel (VV) transitions (iu, ia, ua, au, ai, ui). Time-dependent displacements of the lower lip pellet, four tongue pellets, and a jaw pellet were extracted from each of the VV transitions for each speaker. Also extracted for each VV were F1, F2, and F3. Jaw movement was decoupled from the movement of the other fleshpoints following Westbury (1994). Correlations were then calculated between each flesh point and each of the formants on both the x and the y axis of movement. An analysis of the correlation results indicates that F2 was most highly correlated with articulator movement on both the x and y axes.

5pSC13. Processing of Japanese vowel devoicing and the effect of speech rate. Naomi Ogasawara (Dept. of Linguist., Univ. of Arizona, Douglass 200E, P.O. Box 210028, Tucson, AZ 85721-0028, naomi703@email.arizona.edu)

Vowel devoicing (or reduction) in Japanese is an interface between the phonetic and phonological levels (Vance, 1987; Yoshida, 2002). The phonological rule for vowel reduction defines that high vowels (/i, u/) undergo reduction when between voiceless consonants or between a voiceless consonant and a pause. However, there are some cases of vowel reduction that do not fit the rule. A high vowel adjacent to a voiced consonant sometimes becomes reduced, especially in casual speech (Arai, 1999). Even a nonhigh vowel can undergo reduction (Arai, 1999; Vance, 2004). In this study, the effect of speech rate on the processing of reduced vowels is examined, particularly for vowels that are only likely to be reduced in casual, fast speech. How do listeners recognize vowels that are reduced in fast speech, or the words containing them? A lexical decision experiment showed that listeners found it easier to process fast speech containing reduced vowels than careful speech of isolated words containing them. They also found vowel reduction acceptable for nonhigh vowels in fast speech. This suggests that listeners use knowledge of both phonology and speech rate variability in processing connected speech.

5pSC14. Korean coda cluster simplification: Articulatory study. Minjung Son (Dept. of Linguist., Yale Univ., 370 Temple St., New Haven, CT 06520, and Haskins Labs., 300 George St., #900, New Haven, CT 06511)

Korean exhibits optional coda cluster reduction in lateral-stop (C1C2) sequences. However, this phenomenon has not been tested on the basis of articulatory data. The present study examines articulatory movement using EMMA. For two speakers of Seoul-Korean (e.g., JL and SL), clusters such as /Vlk/ and /Vlp/ followed by /ta/ were elicited at a normal rate along with control tokens (e.g., /Vkta/, /Vpta/, /Vlta/). The stimuli were presented with the relevant word in isolation. Results indicated that neither speaker showed /p/ and /l/ reduction, although there was interspeaker variation in /k/ reduction. While no /k/ reduction occurred in C1C2 for JL, SL demonstrated categorical /k/ reduction in 19% of production, where a longer closure duration of a tongue tip gesture was observed (e.g., possible compensatory lengthening) when it was compared with the control /VltV/; this fails to be addressed in impressionistic transcription and OT analyses [June 11th ICKL, pp. 382–390 (1998); Cho, CLS **35**, 43–57 (1999)]. Lastly, the duration of gestural formation of /k/ in the sequence containing /lk/ was longer than that of /p/ in the sequence containing /lp/, which can be articulatorily supporting evidence to explain more /l/ deletion before /k/ than /p/ in Cho (1999). [Work supported by NIH.]

5pSC15. Intrusive vowels in Cruceno Spanish. Cynthia Kilpatrick, Kathryn McGee (Dept. of Linguist., Univ. of California, San Diego, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108), and James Kirby (Univ. of Chicago, Chicago, IL 60637)

Intrusive vowels are short vowels appearing within consonant clusters that are not treated phonologically like full vowels. Though reported in many languages, few quantititative studies have examined these vowels. Here, an acoustic study of intrusive vowels in obstruent+tap clusters in the Spanish of Santa Cruz, Bolivia is reported. The results support earlier studies in that the intrusive vowel quality resembles that of the following nucleic vowel, rather than a neutral vowel, and intrusive vowels are significantly longer in clusters with voiced obstruents than in those with voiceless ones. However, results do not fully support all previous phonetic descriptions and related theoretical assumptions. In particular, a significant difference is not found in intrusive vowel length based on variables such as place of articulation of the obstruent, quality of the nucleic vowel, or placement of stress in relation to the cluster. In addition, where previous work finds no significance for position in the word, the present work finds that intrusive vowels are significantly longer in word-medial clusters than in word-initial clusters. The data further suggest that the articulation of obstruent+tap does not require an intrusive vowel, as has previously been claimed, as not all obstruent+tap clusters include an intrusive vowel.

5pSC16. Ultrasound study of velar-vowel coarticulation. Sylvie M. Wodzinski and Stefan A. Frisch (Dept. of CSD, Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620)

In velar fronting, the closure location for a velar consonant is moved forward along the palate due to vowel context. This study is a replication and extension of a previous study on velar fronting [Wodzinski and Frisch, J. Acoust. Soc. Am. **114**, 2395 (2003)]. In this study, ten participants

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produced sentences containing monosyllabic words with /k/ onsets and nine different American English vowels. Ultrasound was used to make measurements of lingual-palatal constriction location at the midpoint of stop closure. Participants were recorded using a head-stabilizing apparatus and an acoustically transparent standoff was used between the ultrasound probe and the jaw. Velar closure location was quantified by the angle of elevation from the horizontal axis of the ultrasound probe to the center of the velar closure. The articulatory frontness of the following vowel was quantified using the frequency of $F2$ at the vowel midpoint. A strong negative correlation between velar closure angle and the following vowel $F2$ was seen for all ten participants. The coarticulatory pattern suggests that each velar-vowel combination results in a distinct closure location for the velar. Implications for models of speech production will be discussed.

5pSC17. Introducing diphthongs to the vowel space. Anna Bogacka (Adam Mickiewicz Univ., Poznan, Poland)

The aim of the study is to provide information on a new vowel space of Polish learners of British English, specifically about acoustic properties of diphthongs. Polish does not have diphthongs, though it has vowel plus glide sequences comparable to English rising diphthongs. Polish has neither vowel duration distinctions nor vowels which are qualitatively identical to the ones appearing in English diphthongs. Studying diphthongs offers the possibility of examining the interplay of substitutions of qualitative and quantitative features. Eight dipthongs were taken into account. The conditioning criteria were quality, duration, occurrence of glottal stops, and degree of nasalization. Twenty subjects were recorded reading 61 sentences with diphthongs embedded in real words. The paper discusses challenges in segmentation. Reported are non-native properties of formant and timing relations and systematic differences in alternations applied to: simple vowels versus diphthongs, rising versus centering diphthongs, and initial versus final phases of diphthongs. Specifically, nasal vocalization was found to depend on the presence or absence of a following fricative, but to be independent of a stressed or unstressed position. The finding is explained by resorting to Polish prosody, which is not stress timed. Suggestions for using the findings in second language instruction will also be offered.

5pSC18. Cues to stop place in stop-liquid clusters. Edward Flemming (Dept. of Linguist. and Philosophy, M.I.T., 77 Massachusetts Ave., 32-D808, Cambridge, MA 02139, flemming@mit.edu)

The acoustic correlates of stop place contrasts in prevocalic position have been studied intensively, but much less is known about stop place contrasts in consonant clusters. An understanding of the nature and distribution of cues to contrasts across a variety of contexts is important to the phonological analysis of the distribution of contrasts. This study extends the characterization of stop place contrasts to stop-liquid clusters /br, dr, gr, bl, gl/, based on acoustic analysis of American English. We examine how well established cues relating to burst duration, burst spectrum, and formant transitions generalize to the stop-liquid context. The relative perceptual weight of burst and formant transitions will be assessed through identification of cross-spliced stimuli.

5pSC19. Acoustic analysis of pressed voice. Carlos Ishi, Hiroshi Ishiguro, and Norihiro Hagita (ATR/IRC Labs., 2-2 Hikaridai, Kyoto 619-0288, Japan)

Pressed voice (''rikimi'' in Japanese) is a voice quality related to the vibratory patterns of the vocal folds. It was recently shown that pressed voice carries important paralinguistic information in Japanese, indicating emotional or attitudinal state of the speaker. In the present work, several acoustic features are investigated, aiming for an appropriate acoustic characterization and an automatic detection of pressed voices. Analysis of pressed voice samples extracted from natural conversational speech firstly shows that irregularity in periodicity (such as in creaky and harsh voices) is a common but not a strictly determinant feature of pressed voices. Spectral analysis shows that parameters related to spectral slope are effective to identify part of the pressed voice samples, but fail when vowels are nasalized or double-beating occurs in a glottal cycle. Temporal analyses of pressed and nonpressed creaky voices indicate that diplophonia (simultaneous production of two separate tones, when vocal folds oscillate out of phase) frequently occurs in nonpressed creaky voices, while it rarely appears in pressed ones. Further temporal analysis of EGG (electroglottograph) waveforms of acted speech showed the same trends obtained for natural speech, indicating that information about the absence of diplophonia can potentially be used for pressed voice detection.

5pSC20. Determining Bayesian evidence and decay time estimates in acoustically coupled volumes. Tomislav Jasa (Dept. of Elec. Eng., The University of British Columbia, Vancouver, BC, Canada V6T 1Z2) and Ning Xiang (Rensselear Polytechnic Inst., Troy, NY 12180)

Sound energy decay analysis is an important element in understanding acoustical properties of coupled volumes in architectural acoustics. A Bayesian model formulation using Monte Carlo algorithms [Xiang and Goggans, J. Acoust. Soc. Am. **110** 1415–1424 (2001); Jasa and Xiang, *ibid.* **117**, 3707(A) (2005)] has been applied to estimating both the decay times and decay order required in sound energy decay analysis. The need to focus on model selection in a Bayesian model formulation of sound energy decay analysis was presented by Jasa and Xiang [J. Acoust. Soc. Am. **119**, 3343(A) (2006)] along with a discussion of the limitations of the existing Monte Carlo algorithms used for this purpose. This paper will present some recent algorithms that can overcome the limitations of a Monte Carlo approach to model selection and show how these algorithms can be applied to determining both the proper model and decay time estimates for acoustically coupled rooms.

5pSC21. Acoustic analysis and synthesis of laughter. Toshiaki Haga, Masaaki Honda, and Katsuhiko Shirai (Dept. of Sci. and Eng., Univ. of Waseda, 3-4-1 Okubo, Rm. 55S-609, Shinjuku-ku, Tokyo 169-8555, Japan)

Laughter is an important emotional expression in speech. The production is characterized simply by quasiperiodic glottal opening/closing gesture with relatively constant articulatory configuration. The acoustic consequence, however, is complicated and appears in the temporal characteristics of voiced-unvoiced category and pitch, the spectral envelope, and the segmental duration. In this study, acoustic parameters that characterize laughter were investigated from acoustical and perceptual points of view. Laughter /hahaha/ was selected as a typical laughter from conversational speech recording and the acoustic characteristics were compared with those of normally uttered speech /hahaha/. The acoustic analysis was performed in terms of pitch in average and its intrasegmental pattern, the spectral tilt and formant bandwidth, and the segmental duration of the vowel /a/ and the consonant /h/. Then, perceptually significant acoustic features were investigated by using means of replacement of the acoustic parameters of the normally uttered speech by those of the laughter in speech analysis-synthesis system. The results showed that an increased pitch in average and distinct intrasegmental pitch pattern as well as spectral envelope with broad bandwidths are significant features to characterize the laughter. The laughter synthesis by rule was also investigated by manipulating these acoustic parameters.

5pSC22. A study of manual articulatory feature-based transcription of conversational speech. Karen Livescu (Comput. Sci. and Artificial Intelligence Lab., MIT, Cambridge, MA), Xuemin Chi, Lisa Lavoie (MIT, Cambridge, MA), Ari Bezman (Dartmouth College, Hanover, NH), Nash Borges, and Lisa Yung (Johns Hopkins Univ., Baltimore, MD)

This study investigates the manual labeling of speech, and in particular conversational speech, at the articulatory feature level. A detailed transcription, including subtleties such as overlapping or reduced gestures, is useful for studying the great pronunciation variability in conversational

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speech. This type of labeling also facilitates the testing of automatic feature classifiers, such as those used in articulatory approaches to automatic speech recognition. For this study, approximately 100 utterances drawn from the SWITCHBOARD database have been transcribed using eight articulatory tiers rather than the traditional single phonetic tier. The tiers include: place and degree for up to two constrictions, nasality, glottal state, lip rounding, and vowel quality. Two transcribers have labeled this set of utterances in a multipass strategy, allowing for correction of errors. Preliminary analysis shows a high degree of intertranscriber agreement. Further analysis of the data is being performed to address a number of questions, such as: How quickly and reliably can this type of transcription be done? What are its advantages and disadvantages relative to purely phone-based transcription? What characteristics of the utterances correspond to high or low transcriber agreement? What can be learned from the data regarding articulatory phenomena such as reduction and asynchrony?

5pSC23. Changes in vocal tract resonance during a pitch cycle. Hironori Takemoto, Tatsuya Kitamura (NICT/ATR Cognit. Information Sci. Labs., 2-2-2 Hikaridai, Seika-cho, Soraku-gun Kyoto, 619-0288, Japan, takemoto@atr.jp), and Seiji Adachi (Fraunhofer Inst. for Bldg. Phys., 70569 Stuttgart, Germany)

Vocal tract (VT) resonances during a pitch cycle were examined using area functions of the five Japanese vowels extracted from MRI data of a male subject. The volume-velocity distribution within the VT at each formant was calculated by a transmission line model, as a function of the glottal area. Our previous study showed that in glottal closure condition the fourth formant was provided by the laryngeal cavity (LC) and the other formants by the VT excluding LC, the VT proper (VTp). During the early stage of glottal opening, the fourth formant disappeared, as it had been provided by the closed-tube resonance of LC. After glottal opening, the single volume-velocity node of the first formant moved to the middle pharynx, shortening the effective VT length and thus increasing its frequency. At the other formants, a node appeared at the junction between LC and VTp, and the VT resonance during the glottal opening condition could therefore be approximated by the VTp. This result indicates that the VTp was responsible for most of the VT resonances throughout a period, while LC provided the fourth formant in the glottal closure period. [Work supported by NICT and MEXT KAKENHI of Japan.]

5pSC24. A study on generation of the voice bar in voiced stops. Jianwu Dang (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., Japan), Takayoshi Nakai (Shizuoka Univ., Japan), Yongnan Piao (Japan Adv. Inst. of Sci. and Technol., Japan), and Shinobu Masaki (ATR, Japan)

The voice bar of the voiced stops is generally considered as a radiation sound from the vibration of the laryngeal wall while the mouth is closed. In this study, we investigated the mechanisms involved in generation of the voice bar of the voiced stops by means of acoustic and mechanical measurements and MRI observations for three Japanese speakers. Radiated sounds from the mouth and the nostrils were recorded separately, while vibration of the laryngeal wall was measured by an accelerometer. The correlation of the nostril radiation to the voice bar is higher than that of the laryngeal wall vibration to the voice bar. It implies that the major component of the voice bar is likely radiated from the nostrils but not the laryngeal wall, where the nostrils' radiation is possibly caused by a transvelar coupling of the yielding velum. The authors have proposed a two-layer board model to imitate the transvelar function. To obtain physiological evidences, articulation of stop-vowel sequences was observed by MRI movies. It found that the thickness of the velum varies with the adjacent vowels. Applying this measurement to the two-layer board model, the generation of the voice bar was evaluated by comparing the simulations and observations.

5pSC25. Sound radiations from nostrils for voiced speech and their individuality. Takayoshi Nakai and Masashi Yamamoto (Faculty of Eng., Shizuoka Univ., 3-5-1 Johoku, Hamamatsu, Japan, 432-8561tdtnaka@ipc.shizuoka.ac.jp)

This study is intended research to analyze sound radiations from the nostrils for voiced speech and explore their characteristics. Sound pressure near the nostril, vibration acceleration of the nose, and sound pressure in front of the mouth were observed using a three-channel A-D converter. The ratio of radiations from the nostril and the mouth, RRNM, was observed. Speech materials included nasal sounds /ma/ and /na/; five vowels; and /cvcv/(c=b, d, g, v=a, i, u, e, and o). There are ten male subjects. For initial buzzes, RRNM showed individuality: it is from 0 to −15 dB lower than for /m/. For medial buzzes, RRNM is mostly lower than for initial buzzes. Transitions from buzz to vowel and from vowel to buzz were mostly fast, but their transitions are slow and smooth in some cases. Frequency analyses revealed that radiation from the nostrils consists mainly of fundamental frequency, and the second and third harmonics.

5pSC26. Spectral characteristics of period doubled phonation. Jody Kreiman and Bruce R. Gerratt (Div. of Head/Neck Surgery, UCLA School of Medicine, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Period doubling (also called subharmonic phonation, bifurcated phonation, F0 series) occurs frequently in both pathological and normal phonation. In this kind of phonation, the most obvious acoustic characteristic is that vocal pulses appear in pairs, with alternate pulses differing in period and often in amplitude. Attempts at modeling period doubling by varying period and amplitude did not capture its characteristic quality, leaving the question of which acoustic variables underlie our perception of period doubling. Acoustic analyses of 50 period doubled voices were undertaken to determine what acoustic features might underlie the constant percept associated with period doubling. Every voice examined did show alternation between short/long and large/small glottal pulses. However, alternations in amplitude were not consistently linked to alternations in period. Voices additionally showed a consistent pattern of alterations in pulse shape, so that the A and B pulses consistently differed in H1-H2. Results suggest that the consistent vocal quality associated with period doubling is due to combined alternations in period and spectral tilt. Amplitude alternations appear to be acoustic artifacts of the period and spectral alternations.

5pSC27. On the vocalization of nonspeech sounds: Implicit mechanisms. Rodrigo Segnini (Shimojo Implicit Brain Function Project, JST-ERATO, 3-1 Morinosatowakamiya, Atsugi, Kanagawa, Japan 243-0198, rsegnini@shimojo.jst.go.jp) and Takemi Mochida (NTT Sensory and Motor Res. Group, Atsugi, Kanagawa, Japan 243-0198)

When humans vocalize nonspeech sounds, as when imitating other sound-producing mechanisms, how effective is this vocalization? The human vocal tract is adept at producing speech sounds, allowing verbal communication despite variations in these sounds due to differences in size, shape, and dynamics of this mechanism among individuals. In contrast, nonspeech sounds that were not originally produced by the human vocal tract derive from even more structurally different mechanisms—those of animal calls or machinery sounds, for example. Through experiment, we determined that in imitative tasks of such nonspeech sounds, subjects were intuitively able to produce distinguishing articulatory movements for each sound under different conditions, even when lacking auditory feedback. These results reveal a nontrivial coherence between articulatory events and the intended sound despite the noted differences with the original production mechanism and the resulting acoustic mismatch. The authors would like to express sincere appreciation to Makio Kashino for enriching discussion and support in the realization of this project.

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5pSC28. Basic processes underlying speaking and singing: Preliminary acoustic and aerodynamic comparisons. Amee P. Shah (Dept. of Health Sci., Cleveland State Univ., 2121 Euclid Ave., MC 431-B, Cleveland, OH 44115)

Intuitively, we think speech and singing are different. Perceptually, we hear the difference. Aesthetically, we appreciate the difference. Psychologically, we benefit from the positive effect of singing. The important question is whether this perceived difference is purely perceptual or is it grounded in physiological reality? It is important to identify the difference between speaking and singing if indeed there is one as it can shed light on our theoretical understanding of normal auditory perception and vocal production processes. Addressing this question has practical applications as well, as the findings could yield valuable information for working with clients in the fields of speech pathology and vocal music training. In addressing this question, the present investigation targeted broad areas to identify the differences between speech and singing. Specifically, respiration (the mechanics of breathing), articulation (the movements of the oral musculature), phonation (vocal-fold movement and air pressures), and acoustic attributes (temporal and spectral measures) were compared across speaking and singing. Surprisingly, and somewhat counterintuitively, results showed that speaking and singing are not as different as they are perceived to be. Results of preliminary acoustic and aerodynamic comparisons will be discussed as they help better understand the processes of singing and speaking.

5pSC29. The articulation of consonants in Kinyarwanda's sibilant harmony. Rachel Walker, Dani Byrd, Sungbok Lee, Celeste DeFreitas (Dept. of Linguist., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693, rwalker@usc.edu), and Fidele Mpiranya (Univ. of Chicago, Chicago, IL 60637)

Kinyarwanda's sibilant harmony causes alveolar /s, z/ to become retroflex when preceding a retroflex fricative within a stem. Intervening coronal stops block sibilant harmony, but bilabial and velar consonants are transparent. This study investigates the production of Kinyarwanda's sibilant fricatives and also examines transparent and opaque consonants in the system. Articulatory kinematic data were collected for a native speaker of Kinyarwanda using electromagnetic articulography. This allowed calculation of the mean angle for receivers affixed to the tongue tip and blade over the target consonant intervals. Average mean angle reliably and robustly distinguished alveolar and retroflex fricatives, with alveolars showing a lower tip relative to blade. No significant difference in mean angle was found for /t/ in contexts where it blocked retroflex sibilant harmony versus ones where it preceded an alveolar fricative, confirming that /t/ does not participate in the harmony. However, in contexts where /m/ and /k/ are perceived as transparent to sibilant harmony, their mean tip-blade angle was significantly different from contexts where harmony did not occur. Furthermore, mean angle during transparent /m/ showed no significant difference from mean angle during retroflex fricatives, suggesting that tongue tip-blade angle conducts strongly and systematically through transparent consonants but without perceptible effect. [Work supported by NIH, USC URP.]

5pSC30. Validity of the Nasometer measuring the temporal characteristics of nasalization. Youkyung Bae, David P. Kuehn, and Seunghee Ha (Dept. of Speech and Hearing Sci., Univ. of Illinois at UC, 901 S. Sixth St., Champaign, IL 61820, yousong@uiuc.edu)

Patients with oral-nasal resonance imbalance may present different characteristics not only in the amplitude but also in the temporal aspects of nasalization. Although nasalance has been the most commonly derived measure from the Nasometer (KayPENTAX), the Nasometer also might be useful in providing another important measure, time. Such measurements, however, should be validated in relation to a separate external criterion procedure. This study examined the validity of the Nasometer measuring the temporal characteristics of nasalization. Fourteen adult American English speakers produced three rate-controlled nonsense speech samples: /izinizi/, /azanaza/, and /uzunuzu/. Acoustic data recorded through the Nasometer were compared to those recorded through a standard audio recording setup which served as an external criterion procedure. Four timing variables pertinent to nasalization were measured from the digitized acoustic signals: nasal onset, nasal consonant, nasal offset, and total nasalization. No significant differences were found between the two instrumental conditions in any of four timing variables measured, which suggests the Nasometer can be used as a valid tool to measure the temporal features of nasalization. This study also provided a valid way of segmenting speech tasks using the Nasometer and confirmed the significant effect of different vowel contexts on the temporal characteristics of nasalization.

5pSC31. Pitch determination and sinusoidal modeling for time-varying voiced speech. Masashi Ito and Masafumi Yano (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira. Aoba-ku, Sendai 980-8577, Japan, itojin@riec.tohoku.ac.jp)

A voiced speech signal consists of sinusoidal components of which amplitude and frequency are time varying. Usually the signals are analyzed by assuming that they are stationary within a local analysis window, so they include inevitable errors. To solve this problem, we have proposed a method termed local vector transform (LVT), in which amplitude and phase of the sinusoidal components were approximated by quadratic functions uniquely determined from input spectrum. Two types of experiments were carried out to evaluate the validity of this method. First, time-varying pitch frequencies (F0s) of natural utterances were investigated. F0 determined by LVT greatly improved the accuracy compared to the conventional pitch determination algorithms: cepstrum, autocorrelation, and instantaneous frequency estimation. Second, LVT is applied to the sinusoidal modeling and the amplitude and phase were estimated for every component. The results apparently showed that the signal obtained by LVT was very close to the input. This was quantified by a signal to residual power ratio (S/R). For both of synthesized and naturally uttered speech signals, LVT showed the higher S/R compared to the conventional algorithm. These results indicate that the proposed algorithm is highly effective for pitch determination and sinusoidal modeling for time-varying speech signals.

5pSC32. The effect of stress on consonant vowel (CV) coarticulation: Decoupling of the CV bond. Augustine Agwuele, Harvey Sussman, and Bjorn Lindblom (Dept. of Linguist., Univ. of Texas, Austin, TX 78712)

Two common sources of variation in the speech signal arise from (1) naturally imposed speaking conditions (e.g., stress, tempo), overlaid onto (2) inherent contextual influences. Coarticulatory analyses are affected by both sources of variation, but rarely are they dissociated. The focus of this study was to derive quantitative methods to assess the effect of emphatic vowel stress on preceding stop consonant onsets, apart from, and independent of, anticipatory effects of changing vowel contexts. Three speakers produced V1CV2 tokens, with emphatic stress either on V1 or V2; C $=$/bdg/. There were six V1 and 10 V2 vowel contexts. Novel methods were applied to both standard locus equation scatterplots and multiple regression analyses to isolate F2 midvowel (Hz) and F2 onset (Hz) adjustments as a function of emphatic stress. The analyses show how stress differentially affects the V-midpoints and C-onsets for labial /b/ relative to lingual stops /dg/.

5pSC33. Intrinsic vowel F0 and speech rate. Ian Maddieson (Univ. of New Mexico/Univ. of California, Berkeley Dept. of Linguist., Univ. of New Mexico, MSC 03 2130, Albuquerque, NM 87131)

A relation between vowel height and F0 such that, other things being equal, F0 is a little higher in a high vowel than a low vowel appears to be a phonetic universal. However, the cause of this relationship remains unclear. An insight might be gained from ways in which the relationship changes under transformations of speech rate. Ten speakers of American

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English were recorded saying sentences containing words such as ''bead'' and ''bad'' at fast and slow rates. For the majority of the subjects the intrinsic F0 difference was greater in fast speech, contrary to expectation. The difference cannot be explained by changes in vowel quality between the conditions.

5pSC34. Tongue dorsum location and tongue root retraction in alveolar and palatal clicks in the endangered language N/uu. Amanda L. Miller, Johanna Brugman, Jonathan Howell (Dept. of Linguist., Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853), and Bonny Sands (Northern Arizona Univ., AZ)

Traill (1985) describes Xoo clicks as having velar posterior constriction locations (PCL). Miller *et al.* (to appear) show that the PCLs of Khoekhoe alveolar and palatal clicks are uvular and uvulo-pharyngeal. Ladefoged and Maddieson (1996) discuss clicks involving a uvular posterior release location (PRL) (and an implied velar PCL). An ultrasound investigation of velar and uvular pulmonic stops, alveolar and palatal velar clicks, and palatal uvular clicks, in the endangered language N/uu is presented. Results are for 15 repetitions of each consonant in the u context by four of the remaining speakers. The alveolar click has the tongue dorsum (TD) between the velar and uvular stops, and tongue root retraction (TRR) like the uvulars. The TD in the palatal click is also similar to the velar and uvular stops, but there is no TRR. The TD in the uvular palatal click is at the uvulo-phrayngeal location, and there is TRR similar to the uvular stop, in both the closure and the release. TRR in the alveolar-uvular and palatal-uvulo-pharyngeal clicks explains their phonological patterning with uvulars. The lack of TR retraction in the palatal click explains its patterning with velars. [Work supported by NSF, BCS #0236735.]

5pSC35. Acoustic evidence for the lenis/fortis contrast in California Northern Paiute. Molly Babel (UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720-2650, mbabel@berkeley.edu)

California Northern Paiute (CNP) is spoken in the communities on the eastern slopes of the Sierra Nevada Mountains in northern California. CNP is the southernmost dialect of Northern Paiute, a language belonging to the Western Numic branch of Uto-Aztecan. CNP is a severely endangered language spoken by fewer than ten fluent speakers. CNP has a phonemic contrast that is distinct from that of any other Numic language. CNP has three series of stops: lenis, fortis, and voiced fortis. Previous descriptions of CNP have used only impressionistic judgments to categorize the phoneme series. This paper presents the first acoustic evidence for the lenis/fortis contrast in CNP. Field recordings from five native speakers of CNP were analyzed in this investigation. The lenis/fortis contrast was examined by measuring the closure duration using cues from the waveform and spectrogram in a digital signal processing program. Results of duration measurements support the impressionistic evidence and suggest a voiced and voiceless fortis series in addition to a single lenis series in the phonemic inventory of CNP. This research has implications for the acquisition of categories by second language learners who are first language speakers of a language that does not have such a contrast.

5pSC36. Articulation without acoustics: A case study of Oneida utterance-final forms. Bryan Gick (Dept. of Linguist., University of British Columbia, E270-1866 Main Mall, Vancouver, BC V6T 1Z1, Canada), Karin Michelson (Univ. at Buffalo, Buffalo, NY 14260), and Bosko Radanov (University of British Columbia, Vancouver, BC V6T 1Z3, Canada)

Whether the targets of speech production are primarily articulatory or acoustic has been controversial [Guenther *et al.*, J. Acoust. Soc. Am. **105**(5), 2854–2865 (1999)]. While it has been observed that tongue movements occur even in contexts where they may be acoustically obscured [Browman and Goldstein, LabPhonI: 341–376 (1990); Tiede *et al.*, J. Acoust. Soc. Am. **110**(5), 2, 2657 (2001)], the question of whether natural languages can systematically encode articulations in the absence of acoustic consequences has remained open. A study was conducted to investigate a purported phonological process in the endangered Oneida (Iroquois) language whereby utterance-final forms exhibit fixed sequences having this property. Ultrasound, video, and acoustic data were collected from two native speakers of Oneida in a field setting. Preliminary results indicate that speakers' tongue positions are significantly different during different utterance-final vowel articulations despite these vowels being completely inaudible. Results confirm claims that the Oneida phenomenon is in fact a case of articulation without acoustics, and indicate that it is indeed possible for articulations to be retained even in the systematic absence of auditory goals. [Work supported by NSERC.]

5pSC37. Automatic detection of vowel nasalization using knowledge-based acoustic parameters. Tarun Pruthi and Carol Y. Espy-Wilson (Dept. of Elec. Eng. and Inst. of Systems Res., Univ. of Maryland, College Park, MD 20742, tpruthi@umd.edu)

Researchers in the past have suggested several acoustic correlates of nasalization including extra pole-zero pairs near the first formant ($F1$), a reduction in $F1$ amplitude, and an increase in F1 bandwidth. Even though these correlates have been known for a long time, considerable work is still needed to automate the extraction of acoustic parameters (APs) for nasalization. This work looked at 37 different APs which were pared down to 8 APs based on F statistic obtained by ANOVA. In preliminary experiments, an accuracy of 69.79% has been obtained for the task of discriminating between oral and nasalized vowels on the TIMIT database using a support vector Machine (SVM)-based classifier. The classification was done on a frame basis, and a segment was declared nasalized if more than 30% of the frames were found to be nasalized. Note that all vowels adjacent to nasal consonants were assumed to be nasalized. Thus, the accuracy may actually be higher since some vowels before nasal consonants may not be nasalized. Further, these results were obtained by using a linear kernel in SVMs. We hope the results would improve when a radial basis function kernel is used. [Work supported by Honda and NSF Grant BCS0236707.]

5pSC38. A novel method for the measurement of vowel formant frequencies and its application to the analysis of Japanese vowels over a wide range of ages. Hideki Kasuya (Dept. of Speech, Lang. and Hearing Sci., Intl. Univ. of Health and Welfare, Otawara, Japan 324-8501, kasuyah@snow.ucatv.ne.jp), Kanae Ohta, Hiroki Mori (Utsu-nomiya Univ., Utsunomiya, Japan 321-8585), Toshisada Deguchi, Ghen Ohyama (Tokyo Gakugei Univ., Tokyo, Japan 184-8501), and Yukimasa Muraishi (Univ. of Tokyo, Tokyo, Japan 164-8654)

The first three formant frequencies of the Japanese vowels, measured over a wide range of ages of talkers by one of the authors in 1968, have widely been cited in many speech textbooks and research papers. We decided to update those old data, measured by sound spectrograms, by using a more advanced method. We developed an iterative method for measuring vowel formant frequencies based on an autoregressive with exogenous input (ARX) speech production model that incorporates both voice source and vocal-tract models [J. Acoust. Soc. Jpn. **58**, 386–397(2002)]. In using this method to measure vowel formant frequencies of high-pitched voice, the procedure for determining the initial value of the open quotient (OQ) of the source model was carefully investigated, since it affects significantly the estimation of the lower first formant frequency. Estimated OQs of the vowel /a/ of a talker were found to be stable against various initial values and so were used for analyzing the other vowels of the same talker. Perceptual confirmation of phonetic quality of resynthesized vowels with the fundamental frequency perturbed was also of great importance for accurate measurement. This procedure was successfully applied to 320 male and female talkers, aged 6–24 years.

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5pSC39. Spectral properties of Japanese whispered vowels referred to pitch. Hideaki Konno, Hideo Kanemitsu (Hokkaido Univ. of Education, Hakodate 040-8567, Japan, konno@cc.hokkyodai.ac.jp), Jun Toyama, and Masaru Shimbo (Hokkaido Info. Univ., Ebetsu 069-8585, Japan)

Whispered speech can communicate the same linguistic information as ordinary speech in spite of the great difference of their respective acoustic characteristics. In this respect, whispering is an interesting object for studying speech perception and recognition. In this study, we investigate the spectral properties of five whispered Japanese vowels uttered in isolation and having different pitch. Pitch of ordinary whispered vowels was measured in terms of the manner in which the talkers were listening to pure tones while uttering and adjusting its frequency so that the pitch matched the utterance. For other samples, talkers changed the pitch of utterances to match the given pure tones. Acoustic analyses were carried out on formant frequencies, a spectral tilt, and a peak frequency of wide-ranging spectral shape using second-order LPC method called a global peak. Preliminary results show a tendency of upward shift of $F1$, $F2$, and global peaks, and flattened spectral tilts on overall vowels with increasing pitch. The intended pitch by a talker is not correspondent to a specific formant frequency. Results of pitch comparison tests among samples and pitch matching test with pure tones will also be discussed. [Work supported by MEXT Japan.]

5pSC40. Detecting nonlinearity in voice production systems. Yu Zhang, Chao Tao, and Jack J. Jiang (Dept. of Surgery, Div. of Otolaryngol. Head and Neck Surgery, Univ. of Wisconsin Med. School, Madison, WI 53792-7375)

Many biomedical systems exhibit some form of nonlinear behavior, such as saturation in response to certain inputs. Linear description may not be sufficient to describe the complex nonlinear phenomena, including bifurcation and chaos. The determination of nonlinearity from time series has become an important topic for the analysis and modeling of biomedical systems. In this study, we will apply nonlinear dynamic modeling based on Volterra kernel function to describe the nonlinearity of voice signals. Three nonlinear parameters including nonlinearity-to-linearity amplitude ratio (NLAR), nonlinearity-to-linearity parameter ratio (NLPR), and nonlinear order are employed. Sufficiently lower NLAR and NLPR, and a nonlinear order approaching 1 can be found in periodic voice signals, demonstrating the linear mechanism of periodic voice production. However, aperiodic voices show sufficiently high NLAR and NLAR, and a nonlinear order above 2, indicating the dominant role of nonlinearity in disordered voice production. Furthermore, the effects of the signal length and noises on these three parameters are investigated. These three nonlinear parameters show the robustness of short signal length and noise perturbations, demonstrating their potential applications in measuring the nonlinearity of disordered voice production systems. [Work supported by NIH.]

5pSC41. Concurrent speech disturbs word generation: Semantic, associative, and grammatical processes in picture naming: A pictureword interference study. Masumi Watanabe (Dept. of SLP, Tama Rehabilitation Acad., 1-642-1 Nekabu, Ome, Tokyo, Japan 198-0004), Kazuhiko Kakehi (Cyukyo Univ., Kaizu-cho, Toyota, Japan, 470-0393), Joanne Arciuli, David Vinson, Gabriella Vigliocco (Univ. College London, London WC1H 0AP, UK), and Noriko Iwasaki (UC Davis, Davis, CA 95616)

In pictureword interference (PWI), a picture of an object to be named is presented with a distracter word. Most PWI studies showed semantic interference between a target picture and distracter word (noun). Recently, Vigliocco *et al.* (2005) studied grammatical effect in naming a picture of action. They found that generation of an inflected form of an Italian verb was disturbed by a distracter in the same grammatical class. In experiment 1 of the present study, naming an object with an auditorily presented noun or verb distracter in Japanese was investigated, and grammatical class but not semantic effect was found, which is incongruent with previous findings. Analyzing the distracters revealed that familiarity of the verbs was lower than that of nouns, and half of nouns classified as semantically close were associative (e.g., garage versus car). In experiment 2, familiarity was matched across distracters, and association effect was examined separately. To see time course of interference, stimulus onset asynchrony (SOA) was changed from -300 to $+200$ ms. Again, grammatical effect was observed, and associative as well as semantic effect was found for negative SOAs. Possible factors producing grammatical and associative effects in PWI will be discussed cross-linguistically.

5pSC42. Quantifying the Lombard effect in different background noises. Christian Giguère, Chantal Laroche, Émilie Brault, Julie-Catherine Ste-Marie, Marianne Brosseau-Villeneuve, Bertrand Philippon, and Véronique Vaillancourt (Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON, Canada K1H 8M5)

The Lombard effect of increasing one's vocal effort in the presence of background noise has been quantified by Pearsons *et al.* (1977) as a 0.6-dB increase in speech levels for each dB increase in the background noise up to a ceiling level. Lombard speech has also been investigated in other studies with variable results. This study reports data on the effect of different noises on (1) the slope of the function relating speech levels and noise levels and (2) the spectral structure of speech. Twenty normal-hearing adults were asked to read aloud ten sentences from the hearing in noise test (HINT) to an experimenter seated 1 m away, in quiet and in various noises (white, speech spectrum, babble, and restaurant) presented in the sound field at 60 and 75 dBA. Preliminary findings show that increases in speech levels in natural environmental noises (restaurant and babble) most closely follow Pearsons' data, with a slope of 0.6 dB. In contrast, artificial noises (speech spectrum and white) were associated with lower slopes (0.2 and 0.4 dB, respectively). Findings of this study could be useful in a wider context of modeling the complete speech communication process from talker to listener.

5pSC43. Frequency modulated feedback of ones own voice can cause dysfluence of speech. Xu Yizhong (Res. Inst. for Electron. Sci., Hokkaido Univ., N12-W6 Kita-ku, Sapporo 060-0812, Japan), Akira Toyomura, Sachiko Koyama, and Shinya Kuriki (Hokkaido Univ., Sapporo 060-0812, Japan)

Delayed auditory feedback (DAF) can cause speech dysfluency (e.g., stuttering) in healthy normal subjects. In previous studies, feedback frequency was changed linearly or to a constant value in FAF. In this study, we introduce frequency-modulated feedback of pitch in a sine-wave manner. The modulation depth of pitch ($F0$) of auditory feedback voice was six semitones, and the modulation frequencies of sinewaves were set at 0.05, 0.1, 0.5, 0.9, 2, 4, 6, 8, 10, 12, 14, and 16 Hz. In addition to the FAF experiment, we conducted DAF experiments with delay times of 50, 200, and 400 ms. Participants (18 Chinese and 18 Japanese) were instructed to read sentences as they listened to the altered feedback voice through a headphone under each condition. Speech dysfluency was apparent under all FAF and DAF conditions for both groups. Only when the modulation frequency was 14 Hz did Chinese subjects show significantly larger disturbance than Japanese in FAF. A significant greater disturbance of Chinese in DAF was observed when the delay time was 50 ms. These results indicate that frequency modulation can cause speech dysfluency, as does DAF. [Work supported by the 21st century COE (Center of Excellence) Program.]

5pSC44. An online customizable music retrieval system with a spoken dialogue interface. Sunao Hara, Chiyomi Miyajima, Katsunobu Itou,[a)] , and Kazuya Takeda (Grad. School of Information Sci., Nagoya Univ., Furo-cho, Chikusa-ku, Nagoya, 464-8601, Japan, hara@sp.m.is.nagoya-u.ac.jp)

In this paper, we introduce a spoken language interface for music information retrieval. In response to voice commands, the system searches for a song through an internet music shop or a ''playlist'' stored in the local PC; the system then plays it. To cope with the almost unlimited size

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of the vocabulary, a remote server program with which a user can customize their recognition grammar and dictionary is implemented. When a user selects favorite artists, the server program automatically generates a minimal set of recognition grammars and a dictionary. The system then sends them to the interface program. Therefore, on average, the vocabulary is less than 1000 words for each user. To perform a field test of the system, we implemented a speech collection capability, whereby speech utterances are compressed in free lossless audio codec (FLAC) format and are sent back to the server program with dialogue logs. Currently, the system is available to the public for experimental use. More than 100 users are involved in field testing. In our presentation, we will report details of the system and the results of field tests, which include motorcycle environments. [a] Currently at Graduate School of Computer and Information Sciences, Hosei University, Tokyo 184-8584, Japan.

5pSC45. Database construction and analysis of user speech with real environment spoken guidance systems. Hiromichi Kawanami, Manabu Kida, Tobias Cincarek, and Kiyohiro Shikano (Grad. School of Information Sci., Nara Inst. of Sci. and Technol., 8916-5 Takayama-cho, Ikoma-shi, Nara, 630-0101, Japan, kawanami@is.naist.jp)

A spoken guidance information system ''Takemaru-kun'' has developed and been operated in a public city center in Ikoma city, Nara, Japan since November 2002. The system employs the example-based response selection strategy and answers to users' questions about the hall facility, the city, sightseeing, etc. with CG agent animation. Following the success of the system, two new spoken systems ''Kita-chan'' and ''Kita-chan robot'' were developed and were set on a railway station since March 2006. ''Kita-chan robot'' is being operated with a robotlike agent. All user speech of the systems is recorded and two year's data of ''Takemaru-kun'' and one month's data of ''Kita-chan'' systems are manually transcribed and labeled by age and gender. Proper answers are also given to each utterance. In this paper, details of the three user speech databases are described and analytical results concerning numbers of inputs and response accuracy and age group and gender are reported.

5pSC46. A speech communication environment using open source software library for active sound image control. Yuichiro Kitashima, Kazuhiro Kondo, and Kiyoshi Nakagawa (Yamagata Univ., Jounan 4-3-16, Yonezawa, 992-8510, Yamagata, Japan)

This paper describes a three-dimensional (3-D) conference system using an open source software library on conventional PCs. We will attempt to use both 3-D graphics and audio to construct a virtual conference environment for effective communication between remote parties. A rough prototype system was developed using OpenGL and OpenAL. The system uses local files for voice output, whose image location is rendered according to the user input. We initially evaluated the perceived sound image location accuracy of the rendered sound image using the prototype. Users were asked to identify the location of the rendered sound image from among four choices: front, back, left, and right. The users were able to identify the left and right images correctly at virtually 100%, but the front and back identification were lower than 10% for some sounds, particularly male speech. We plan to implement audio-streaming functions to achieve real-time audio conferencing and evaluate the benefits of 3-D audio for conferences. We would also like to implement HRTF (head-related transfer function) and RTF (room transfer function) for improved audio image localization, especially to achieve accurate elevation perception.

5pSC47. Khmu voice register: Acoustic analysis and perceptual experiments. Arthur S. Abramson, Patrick W. Nye (Haskins Labs., 300 George St., New Haven, CT 06511, arthur.abramson@uconn.edu), and Theraphan L-Thongkum (Chulalongkorn Univ., Bangkok 10330, Thailand)

Khmu has phonological voice registers, i.e., bundles of phonetic properties pertinent to the syllable. Auditory observations claim clear voice and high pitch for register 1 and breathy voice and low pitch for register 2. Although Suai registers were in flux [Abramson *et al.*, Phonetica **61**, 147–171 (2004)], it was understood that Khmu Rawk had a stable distinction. Words were recorded by 25 native speakers. Acoustic analysis yielded F0 and overall amplitude contours, frequencies of F1 and F2 in quasi-steady states of the vowels, relative intensities of higher harmonics to that of the first harmonic, and vowel durations. When circumstances caused early attention to perception testing, the words of only eight speakers had been analyzed. Since the only significant factor that had emerged by then was F0 contour, the synthetic stimuli were made with a series of seven contours. The labeling by 32 native speakers yielded two categories, demonstrating the sufficiency of F0 as an acoustic cue. The completed acoustic analysis showed a significant effect of the harmonic ratios for the women, suggesting a conservative bias. The language is shifting toward tonality. Further perception testing must be done for phonation-type effects. [Work supported by NIH and the Thailand Research Fund.]

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Session 5pUW

Underwater Acoustics and Acoustical Oceanography: Scattering, Reverberation, and Noise

Hiroyuki Hachiya, Cochair
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Dezhang Chu, Cochair
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Chair's Introduction—1:00

Contributed Papers

1:05

5pUW1. Modeling scattering from simple three-dimensional bathymetric features using wave-field superposition. John A. Fawcett (DRDC Atlantic, P.O. Box 1012, Dartmouth, N.S., Canada, B2Y 3Z7, john.fawcett@drdc-rddc.gc.ca)

In the method of wave-field superposition a set of point sources inside or outside of a scattering region can be used to represent the exterior or interior acoustic fields. A method is presented that represents a simple bathymetric feature as a closed scattering region in an otherwise flat water-seabed environment. In particular, the feature is taken to be azimuthally symmetric about the vertical axis, thus allowing the three-dimensional scattering problem to be solved in terms of a set of two-dimensional problems. These problems are solved using the method of wave-field superposition. Numerical examples are presented. As well, a benchmark solution in the case that the surrounding medium has only a density jump at the seabed is derived and compared with the more general numerical approach.

1:20

5pUW2. Finite-difference modeling of scattering from shallow buried objects in sea floor. Inger Helene Moen and Hefeng Dong (Acoust. Group, Dept. of Electron. and Telecommunications, NTNU, O.S. Bragstads plass 2B, 7091 Trondheim, Norway)

An object buried in the sea floor positioned close to the water/sediment surface can, when exposed to an incident wave, be a source of interface wave generation. The amplitude of a surface wave tends to dominate at greater distances, and surface waves can thus be a viable tool for detecting shallow buried objects. In this paper, the 2-D velocity-stress finite-difference formulation of Virieux [Geophysics **51**, 889–901 (1986)] is adopted to model the wave propagation from a point source located above the sea floor in an open ocean environment. The wave responses from an object situated below the surface are computed and analyzed, with respect to various object depths, properties, and sizes. A second consideration of the paper is the analysis of the artificial reflections from the model boundaries. The perfectly matched layer absorbing boundaries is a widely used technique that has been shown to be very efficient and has therefore been implemented in order to reduce the grid reflections. The numerical results show that the amplitude of the surface wave decreases with the depth of the object, and that interface waves can be of great importance for detection and identification of shallow buried objects in the sea floor.

1:35

5pUW3. Time-frequency analysis of backscattered echoes from absorbing spherical shells. Hui Ou, Xudong Wang, Vassilis L. Syrmos (Dept. of Elec. Eng., Univ. of Hawaii at Manoa, Honolulu, HI 96822), and John S. AllenIII (Univ. of Hawaii at Manoa, Honolulu, HI 96822)

The backscattering of tone bursts by spherical shells immersed in water has been investigated by several researchers. For a very thin shell, the backscattering specular echo is followed by enhanced echoes. The mechanism of these enhanced echoes has been attributed to a subsonic surface guided wave from recent ray synthesis analysis [P. L. Marston and N. H. Sun, J. Acoust. Soc. Am. **96**, 1862–1874 (1992)]. In this study, time-frequency analysis is performed on the scattering signal. A discrete Wigner distribution function is employed to express the information content in the scattered sound in relation to the size and material of the shell [G. C. Gaunaurd and H. C. Strifors, IEEE **84**, 1231–1248 (1996)]. The simulation results support that the time delay between the echoes is determined by the outer radius of the shell as well as the shell material (with or without absorption), or the Rayleigh velocity of the shell material. Furthermore, the backscattered enhancement occurs at different ranges in frequency domain due to various shell materials, which provide a potential method to determine the shell material properties. The relationship of this work to target discrimination methods for sonar applications is highlighted. [Work supported by ONR.]

1:50

5pUW4. Numerical and experimental investigations of transformations of near-field to far-field scattering measurements. John A. Fawcett, Juri Sildam, and Mark V. Trevorrow (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS, Canada, B2Y 3Z7, john.fawcett@drdc-rddc.gc.ca)

Due to the size of a target and measurement constraints, the measurements of monostatic acoustic scattering are often made in the near field of the target. For many applications, it is an estimate of the far-field target scattering strength, which is desired. In this presentation some model-based inversion techniques, which can be used to predict far-field monostatic target strengths as a function of aspect angle, are derived. In particular, a set of near-field measurements is used in an inversion scheme to compute coefficients for either a linear array of point scatterers or for a generalized Kirchhoff model of the target. The resulting model is then used to predict the far field scattering amplitudes. Simulated and experimental data are used to illustrate the accuracy of the proposed methods.

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2:05

5pUW5. Scattering in swell-contaminated seas: A semiempirical approach. Roger C. Gauss and Joseph M. Fialkowski (Naval Res. Lab., Code 7144, Washington, DC 20375-5350)

Acoustic scattering from the undersea air-sea interface is driven by its surface roughness. In modeling this scattering, 2-D spectral models such as the Pierson-Moskowitz model are used to represent the roughness contribution. However, these spectral models typically assume fully developed wind-driven seas. In practice, seas are not always fully developed and swell is often present. In general, scattering depends on the full roughness spectrum and so can be sensitive to its peak behavior. This paper describes low-to-midfrequency (<10 kHz) bistatic scattering strength and coherent surface-loss models that are acoustically constrained by backscatter data and oceanographically constrained by coincidentally measured wave spectra. To match the latter, a semiempirical surface-wave spectral model is developed whose key parameters (strength, exponent, and peak) are expressed solely in terms of significant wave height and wind speed. It is shown that including the non-wind-wave contributions is essential to accurately modeling interface scattering in general at low frequencies, and forward scattering at all frequencies. [Work supported by ONR and PEO C4I & Space (PMW-180).]

2:20

5pUW6. Modeling reverberation time series for shallow water clutter environments. Kevin D. LePage (Naval Res. Lab., Code 7144, 4555 Overlook Ave. SW, Washington, DC 20375), Charles W. Holland (The Penn State Univ., State College, PA 16801), and Peter Neumann (Planning Systems Inc., Reston, VA 20191)

The phenomenon of clutter in shallow water environments can be modeled from a variety of viewpoints. One approach we have taken is to model reverberation time series for heterogeneous environments with a variety of scattering mechanisms to see if some of the characteristics of resulting estimates model the characteristics of experimentally observed clutter. Here we show reverberation time series estimates for the Straits of Sicily that seem to capture some of the clutter behavior observed during the Boundary 04 experiment conducted in conjunction with the NATO Undersea Research Centre. Of particular interest is the appearance of coherent time-frequency striation patterns in both the modeled and observed reverberation, with compact time supported but broadband clutter features superimposed by the presence of environmental discontinuities. [Work supported by ONR Phase II STTR N00014-04-C-0399.]

2:35

5pUW7. Mechanism for sea bottom reverberation in long range sonar in a range-dependent continental shelf environment. Ameya Galinde and Purnima Ratilal (Northeastern Univ., 409 Dana Res. Ctr., Boston, MA 02115)

The statistically stationary background reverberation in long-range sonar data acquired in the New Jersey Continental shelf during the ONR 2003 Main Acoustic Clutter Experiment is mainly due to seafloor scattering. We compare three mechanisms or models for scattering from the sea bottom in a range-dependent waveguide with the experimental data. They are the Rayleigh-Born volume scattering model, a rough surface scattering model, and an empirical Lambertian model. Each of these models has a different decay rate as a function of range and different frequency dependence. These models are compared to the measured background reverberation as a function of range from the sonar system and at various frequency bandwidths from 300 to 1500 Hz to determine which mechanism or model for seafloor scattering best describes the data.

2:50

5pUW8. "Through-the-sensor" monostatic bottom loss and bottom backscatter measurements at mid-frequencies. Martin Barlett and Walter Brown (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713, barlett@arlut.utexas.edu)

Determinations of bottom backscatter and bottom loss parameters for the seafloor at mid-frequencies are of interest for understanding the performance of modern tactical sonar systems. A variety of techniques exist for measuring bottom scattering properties, but many of these methods require significant resources and/or time to perform. Over the last several years, a through-the-sensor approach for making bottom backscatter measurements using standard tactical sonars has been developed. Recently, these techniques have been extended to monostatic bottom loss estimates. In this presentation, a description of the overall approach to these measurements, advantages and limitations of these techniques, and examples of through-the-sensor bottom backscatter and bottom loss measurements made from data collected through a tactical sonar system will be given.

3:05–3:20 Break

3:20

5pUW9. Validating physics-based clutter from seafloor roughness and biologic scattering. Juan Arvelo, Charles Monjo, Bruce Newhall (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099), Adam Frankel (Marine Acoust. Inc., Annapolis, MD 21401), Richard Love (Fort Worth, TX 76179), and William T. Ellison (Marine Acoust. Inc., Litchfield, CT 06759)

Two physics-based clutter models were developed for comparison and to study the merits and limitations of both approaches. The stochastic approach [Newhall and Arvelo, J. Acoust. Soc. Am. **118**, 2041 (2005)] makes use of perturbation theory, to relate the mean seafloor scattering strength to the mean surface wavenumber spectrum, and the extrapolation of this spectrum to the Bragg wavenumber to generate the clutter distribution from the seafloor roughness. The semi-deterministic approach [Monjo and Arvelo, J. Acoust. Soc. Am. **118**, 2041 (2005)] is based on 2-D power-law fractal realizations of the ocean floor, coherent modal addition, and the microscale bathymetric slopes. Biologic scattering was later included where spatial distribution and nonstationarity of schools of fish and individual marine mammals are realistically represented [Frankel *et al.* J. Acoust. Soc. Am. **119**, 3437 (2006)] through individual-based modeling of their movements, school size, and target strengths. Clutter distributions from both approaches are compared against each other, for selected test cases, and against clutter measurements. [This effort was funded by the Maritime Sensing Program of the Office of Naval Research (ONR Code MS 321).]

3:35

5pUW10. Ocean bottom scattering: characterization with chirp sonar. Jorge Quijano, Lisa Zurk (Portland State Univ., P.O. Box 751-ECE, Portland, OR 97207, zurkl@cecs.pdx.edu), Altan Turgut (Naval Res. Lab/DC, Washington, DC 20375), and D. J. Tang (Appl. Phys. Lab., Seattle, WA 98105)

Shallow water acoustic propagation in the mid-frequency range ($\sim 1-10$ kHz) is strongly affected by scattering from the ocean bottom. In general, this scattering is a combination of rough surface scattering from the water-sediment interface and sediment layers, as well as volume scattering from embedded inhomogeneities. It is necessary to develop an understanding of the spatial and temporal characteristics of the acoustic field to optimize Navy sonar performance. Although there has been a growth of research activity in this area, the validity of different modeling approaches still remains an open research question. One promising tool for bottom sediment characterization is a chirp sonar. These sonars can produce high-quality data in which time of arrival is often associated with location of subbottom sediment layers to produce sediment profiles. However, features of the acoustic scattering such as multiple scattering or particle resonances can have complicated signatures that are not straightforward to

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interpret. In this work, the effect of these signatures on the chirp sonar data is studied through application of a numerical FDTD model applied to a variety of ocean bottom geometries. Results obtained from the chirp sonar system are presented and compared with ocean bottom scattering models.

3:50

5pUW11. Using the ultrasound standing wave in vibroacoustography for surface roughness imaging applications. Farid G. Mitri and Mostafa Fatemi (Dept. of Physiol., and Biomed. Eng., Ultrasound Res. Lab., Mayo Clinic College of Medicine, 200 First St. SW, Rochester MN 55905)

The surface roughness (of about 20 microns) of an acrylic block immersed in water was measured using the acoustic emission pressure field (in the kHz range) induced by the dynamic radiation force of ultrasound (2 MHz). The dynamic radiation force is produced by mixing two beams of different frequencies to excite the probed object. The acoustic emission is a result of object vibration at the difference frequency of the incident ultrasound beams and is detected by a low-frequency hydrophone placed nearby the block. Due to the presence of the block, an ultrasound standing wave field is generated as a result of multiple reflections between the transducer and the block's surface. The acoustic emission by the block's surface varies according to its relative position to the transducer. This variation is used to form an image of the surface roughness at two difference frequencies $\Delta_f = 9.9$ and 22.4 kHz. To have a one-to-one map between the surface roughness and the image, the surface variation has to lie within the distance from a maximum to its nearest minimum in the standing wave. Images obtained here demonstrate that vibroacoustography may be used as a powerful tool for the nondestructive inspection and imaging of surface roughness [Mitri *et al.*, Appl. Phys. Lett. **88**, 234105 (2006)]. [Work supported by NIH.]

4:05

5pUW12. Tower-based breaking wave noise measurements. Steven L. Means and Jeffrey A. Schindall (Naval Res. Lab., Code 7120, 4555 Overlook Ave. SW, Washington, DC 20375, means@wave.nrl.navy.mil)

A tower-based sea surface noise experiment began collecting data January 2006 in the shallow waters (25 m) approximately 75 km off the coast of Savannah, Georgia. A 32-phone, three nested-aperture, vertical hydrophone array was deployed nominally 100 m from a Navy tower that stands 50 m above the water surface. A high-resolution video camera was mounted near the top of the tower along with a dual-polarized marine-band radar to record the location, size, and lifetime of the surface expressions of the breaking waves above of the array. The array is cabled back to the tower for power and signal collection. The tower is microwave-linked to shore for internet-based control and data retrieval. To date, measurements have been made in wind speeds ranging from 5–21 m/s and wave heights of 1–3.4 m over a 6-month period. Empirical relationships between the time-frequency structures of the generated noise, obtained on the endfire beam of the array, and the size of the surface expressions of the breaking waves will be given for a range of environmental conditions. [Work supported by ONR base funding at NRL.]

4:20

5pUW13. Numerical simulation of noise generated by wave breaking in a tank. Xuemei Chen, Steven L. Means, Bill G. Szymczak (Acoust. Div., Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375, xchen@wave.nrl.navy.mil), and Joel C. W. Rogers (Brooklyn Polytechnic Univ., Brooklyn, NY)

A coupled hydro-acoustic source model in relating the physical parameters of wave breaking to the source quantities has been developed. The physical processes of wave formation and breaking are modeled using a generalized hydrodynamics formulation with the initial wave profile calculated by boundary integral method. Hydrodynamic parameters, such as pressure variations and air cavity shapes, etc., will be provided through the simulation. In the acoustic simulation, an algorithm has been developed in handling wave propagation in irregular regions, such as the bubbly liquid generated by wave breaking. To validate the modeling, an experiment on wave formation, propagation, and breaking is carried out in a wave tank. The waves are generated mechanically with a computer controlled vertically oscillating wedge. Favorable agreement is found upon comparing prediction and measurement. [Work supported by ONR.]

4:35

5pUW14. Long-term trends in ocean noise. D. Benjamin Reeder, Rommel Pucan, and Curtis A. Collins (Naval Postgraduate School, 833 Dyer Rd., SP-222 Bldg. 232, Monterey, CA 93943)

The effect of ocean noise, particularly from anthropogenic sources such as Navy sonar systems, on marine mammal hearing, behavior, and reproduction is of great concern. However, relatively few studies exist to make a definitive conclusion regarding rising ambient noise levels, due largely to the dearth of information over sufficiently long time periods for a sufficient number of geographical locations. A few studies have estimated that ocean noise is rising at a rate of approximately 3–5 dB per decade at some frequencies in some locations. This study is based upon ambient noise measurements from U.S. Navy sonobuoys deployed by operational units between 1993 and 2004. Data and analysis are presented for three areas of the ocean—the Northwestern Atlantic Ocean, the Greenland-Iceland-United Kingdom (GIUK) gap, and the Mediterranean Sea. Analysis of this data demonstrates a less significant increase, and in some cases, a decrease, in ocean noise for these three ocean areas at frequencies between 50 and 2000 Hz. Ocean noise levels in this study are comparable to measured ocean noise levels reported in the literature during the 1960's.

4:50

5pUW15. Vessel noise measurements underwater in the Haro Strait, WA. Scott Veirs (Beam Reach Marine Sci. and Sustainability School, 6537 16th St. NE, Seattle, WA 98115, scott@beamreach.org) and Val Veirs (Colorado College, Colorado Springs, CO 80903)

In the recent listing of southern resident orcas as endangered under the Endangered Species Act, disturbance by vessels is cited as a risk factor for the population and a factor in the designation of critical habitat. To help determine whether vessel noise underwater is disturbing orcas as they use sound to navigate, forage, or communicate, the source level of common large vessels transiting Haro Strait was measured. Since large vessels are the dominant source of underwater sound for the southern residents (based on water volume ensonified and duration of exposure at a given receive level), it is convenient that all commercial vessels longer than 65 ft are now required to broadcast their location, speed, and other data in real-time. The automatic identification system (AIS) broadcasts data every 2 s via VHF that can be received and decoded with specialized radio receivers that are now coming on the market. AIS was used to log range, speed, and relative bearing while simultaneous recordings were made of the received sound pressure levels using calibrated hydrophones and photographs were taken. From these observations, the broadband and spectrum source level of each vessel was calculated as a function of bearing relative to vessel axis.

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