

Session 2aAAa

Architectural Acoustics: Methods to Quantify Opera House Acoustics I: Basic Studies

Roberto Pompoli, Chair

Dipartimento di Ingegneria, Universita di Ferrara, Via G Saragat 1, 44100 Ferrara, Italy

Chair's Introduction—7:30

Invited Papers

7:40

2aAAa1. Theory of acoustic design of opera house and a design proposal. Yoichi Ando (1-10-27 Yamanokami, Kumamoto 862-0915, Japan)

First of all, the theory of subjective preference for sound fields based on the model of auditory-brain system is briefly mentioned. It consists of the temporal factors and spatial factors associated with the left and right cerebral hemispheres, respectively. The temporal criteria are the initial time delay gap between the direct sound and the first Reflection (Dt1) and the subsequent reverberation time (Tsub). These preferred conditions are related to the minimum value of effective duration of the running autocorrelation function of source signals (te)min. The spatial criteria are binaural listening level (LL) and the IACC, which may be extracted from the interaural crosscorrelation function. In the opera house, there are two different kind of sound sources, i.e., the vocal source of relatively short values of (te)min in the stage and the orchestra music of long values of (te)min in the pit. For these sources, a proposal is made here.

8:00

2aAAa2. Magnetoencephalographic responses in relation to temporal and spatial factors of sound fields. Yoshiharu Soeta, Seiji Nakagawa, Mitsuo Tonoike (Life Electron. Lab., Natl. Inst. of Adv. Industrial Sci. and Technol. (AIST), Midorigaoka, Ikeda, Osaka 563-8577, Japan), Takuya Hotehama, and Yoichi Ando (Kobe Univ., Rokkodai, Nada, Kobe 657-8501, Japan)

To establish the guidelines based on brain functions for designing sound fields such as a concert hall and an opera house, the activities of the human brain to the temporal and spatial factors of the sound field have been investigated using magnetoencephalography (MEG). MEG is a noninvasive technique for investigating neuronal activity in human brain. First of all, the auditory evoked responses in change of the magnitude of the interaural cross-correlation (IACC) were analyzed. IACC is one of the spatial factors, which has great influence on the degree of subjective preference and diffuseness for sound fields. The results indicated that the peak amplitude of N1m, which was found over the left and right temporal lobes around 100 ms after the stimulus onset, decreased with increasing the IACC. Second, the responses corresponding to subjective preference for one of the typical temporal factors, i.e., the initial delay gap between a direct sound and the first reflection, were investigated. The results showed that the effective duration of the autocorrelation function of MEG between 8 and 13 Hz became longer during presentations of a preferred stimulus. These results indicate that the brain may be relaxed, and repeat a similar temporal rhythm under preferred sound fields.

8:20

2aAAa3. Acoustical measurements of sound fields between the stage and the orchestra pit inside an historical opera house. Shin-ichi Sato, Nicola Prodi (Eng. Dept., Univ. of Ferrara, via Saragat 1, 44100 Ferrara, Italy), and Hiroyuki Sakai (Kobe Univ., Rokkodai, Nada, Kobe 657-8501, Japan)

To clarify the relationship of the sound fields between the stage and the orchestra pit, we conducted acoustical measurements in a typical historical opera house, the Teatro Comunale of Ferrara, Italy. Orthogonal factors based on the theory of subjective preference and other related factors were analyzed. First, the sound fields for a singer on the stage in relation to the musicians in the pit were analyzed. And then, the sound fields for performers in the pit in relation to the singers on the stage were considered. Because physical factors vary depending on the location of the sound source, performers can move on the stage or in the pit to find the preferred sound field.

Contributed Papers

8:40

2aAAa4. A scale value for the balance between stage and pit and inside an historical opera house. Nicola Prodi and Sylvia Velecka (Dipartimento di Ingegneria, Universit di Ferrara, via Saragat1, Ferrara, Italy)

Despite its recognized importance, the balance between the singer and the orchestra inside an opera house has received minor attention in the past. In fact, after the fundamental work of Meyer [J. Meyer, "Some problems of opera house acoustics," Proceedings of 12th I.C.A., Vancou-

ver, 1986, pp. 13–18], who explained why the solo singing voice can compete with the orchestra, only partial results were reported on this perceived attribute. In this work a reference scale to assess the balance inside an historical opera house is achieved by means of listening tests inside a controlled room. Two scaling experiments were performed based on the acoustical data measured inside an historical opera house, the Teatro Comunale di Ferrara, Italy. By doing so all of the relevant acoustical characteristics of a typical Italian-style opera house could be exactly reproduced. Acceptable values do not differ much in the stalls and in the boxes and are within 2 dB(A) to +2.3 dB(A). The transfer of the findings to other types of opera houses is discussed, too.

8:55

2aAAa5. Analysis of masking effects on speech intelligibility with respect to moving sound stimulus. Chiung Yao Chen (168, Gifeng E. Rd., Wufeng, Taichung County 413, Taiwan, R.O.C., chychen@mail.cyut.edu.tw)

The purpose of this study is to compare the disturbed degree of speech by an immovable noise source and an apparent moving one (AMN). In the study of the sound localization, we found that source-directional sensitivity (SDS) well associates with the magnitude of interaural cross correlation (IACC). Ando *et al.* [Y. Ando, S. H. Kang, and H. Nagamatsu, J. Acoust. Soc. Jpn. (E) **8**, 183–190 (1987)] reported that potential correlation between left and right inferior colliculus at auditory path in the brain is in harmony with the correlation function of amplitude input into two ear-canal entrances. We assume that the degree of disturbance under the apparent moving noisy source is probably different from that being installed in front of us within a constant distance in a free field (no reflection). Then, we found there is a different influence on speech intelligibility between a moving and a fixed source generated by 1/3-octave narrow-band noise with the center frequency 2 kHz. However, the reasons for the moving speed and the masking effects on speech intelligibility were uncertain.

9:10

2aAAa6. Multi-dimensional analysis of subjective acoustical ratings and acoustical measures in existing concert halls. Toshiyuki Okano (Takenaka Res. and Development Inst., 1-5-1, Ohtsuka, Inzai, Chiba 270-1395, Japan, okano.toshiyuki@takenaka.co.jp)

Correlations between subjective acoustical ratings and hall-averaged values of acoustical measures are studied among existing worldwide major concert halls. It was shown that the classified acoustical ratings by Beranek [*Concert and Opera Halls, How They Sound* (ASA, 1996)] are discriminated correctly by combining binaural quality index (BQI) with some other acoustical measures. BQI is determined by the arithmetic average of inter-aural cross correlation coefficient in three octave bands of 500, 1000, and 2000 Hz, subtracted from unity, calculated from the early 80-ms part of binaural impulse response. Considering that the upper limit value of BQI not to cause disturbing image shift is approximately 0.85 at individual seat [Okano, J. Acoust. Soc. Am. 2219–2230 (2000)], the values of 0.6 or higher in hall averaged value of BQI, 0.85 or smaller in

individual seat value of BQI, and approximately 5 dB or higher in strength factor at middle frequencies are proposed as design objectives to attain a high acoustical quality. It should be provided that other acoustical measures are also optimized. These target values will be very effective in studying room shape of halls, using scale models or computer models.

10:20

2aAAa7. Microperforated stretched ceilings basics and applications. Christian Nocke and Catja Hilge (Akustikbuero Oldenburg, Alte Raad 20a, D-26127 Oldenburg, Germany, nocke@akustikbuero.info)

In 2001 a new product based on the theory of microperforated sound absorbers has been introduced. The thin foil of a stretched ceiling has been microperforated. Stretched-membrane ceilings have been a popular product with regard to modern architecture worldwide for more than 25 years. However, so far only optical and other aspects of the product were generally of interest. A brief introduction of the underlying theory of microperforated sound absorbers according to D.-Y. Maa will be given. Four geometrical parameters (diameter of perforation, distance between holes, thickness of panel/foil, and backing air volume) fully determine the sound-absorption coefficient of these sound absorbers. Results of independent qualification measurements of various assemblies will be shown. The agreement between theoretical predictions and measurements from the reverberation chamber is very high. Finally, different applications in a church, a sports hall, and other rooms will be presented. Again, the comparison between theoretical design and final result show a very high degree of agreement.

9:40

2aAAa8. Lateral efficiency in small auditoriums. Henrik Moller and Timo Peltonen (Akukon Oy Consulting Engineers, Kornetintie 4 A, FIN-00380 Helsinki, Finland, henrik.muller@akukon.fi)

The paper will examine the measured lateral efficiency parameters measured in small halls, that is, halls with less than 750 seats. Comparisons between measured LF and IACC parameters will be presented as well as a study of the correlation between the lateral efficiency parameters and the width and other geometrical attributes of the halls. The measurements were done as part of a larger survey of the Finnish halls that were built between 1980 and 2000.

TUESDAY MORNING, 25 MAY 2004

ROYAL BALLROOM A, 10:15 TO 11:55 A.M.

Session 2aAAb

Architectural Acoustics: Methods to Quantify Opera House Acoustics II: Singer Acoustics

Yoichi Ando, Chair

1-10-27 Yamanokami, Kumamoto 860-0915, Japan

Chair's Introduction—10:15

Invited Papers

10:20

2aAAb1. Singer's preferred acoustic condition in performance in an opera house and self-perception of the singer's voice. Dennis Noson (BRC Acoust., 1741 1st Ave. S., Ste. 401, Seattle, WA 98134, noson@alum.mit.edu), Kosuke Kato,^{a)} and Yoichi Ando^{b)} (Kobe Univ., Kobe, Japan)

Solo singers have been shown to over estimate the relative sound pressure level of a delayed, external reproduction of their own voice, singing single syllables, which, in turn, appears to influence the preferred delay of simulated stage reflections [Noson, Ph.D. thesis, Kobe University, 2003]. Bone conduction is thought to be one factor separating singer versus instrumental performer judgments of stage acoustics. Using a parameter derived from the vocal signal autocorrelation function (ACF envelope), the changes in singer

preference for delayed reflections is primarily explained by the ACF parameter, rather than internal bone conduction. An auditory model of a singer's preferred reflection delay is proposed, combining the effects of acoustical environment (reflection amplitude), bone conduction, and performer vocal overestimate, which may be applied to the acoustic design of reflecting elements in both upstage and forestage environments of opera stages. For example, soloists who characteristically underestimate external voice levels (or overestimate their own voice) should be provided shorter distances to reflective panels—irrespective of their singing style. Adjustable elements can be deployed to adapt opera houses intended for bel canto style performances to other styles. Additional examples will also be discussed. ^{a)}Now at Kumamoto Univ., Kumamoto, Japan. ^{b)}Now at: 1-10-27 Yamano Kami, Kumamoto, Japan.

10:40

2aAAb2. Blending vocal music with a given sound field due to the characteristics of the running autocorrelation function of singing voices. Kosuke Kato (Grad. School of Sci. and Technol., Kumamoto Univ., 2-39-1 Kurokami, Kumamoto, Japan, 031d9203@gstt.stud.kumamoto-u.ac.jp), Kenji Fujii (Yoshimasa Electron., Inc., Shibuya-ku, Tokyo, Japan), Keiji Kawai (Kumamoto Univ., Kumamoto, Japan), Yoichi Ando^{a)} (Kobe Univ., Kobe, Japan), and Takashi Yano (Kumamoto Univ., Kumamoto, Japan)

This is a study to meet music and the opera house acoustics. It is said that singers adjust their interpretation style according to the acoustical condition of the sound field in a room. However, this mechanism of blending of musical performance with the sound field is unknown. In order to obtain a method of performance blending of opera house acoustics, we attempted to develop evaluation criteria for a singing voice in terms of the minimum value of the effective duration of the running autocorrelation function (r -ACF), $(te)_{min}$, of sound source signals. This temporal factor has shown to have close correlation with the subjective response of both listeners and performers to sound fields [Y. Ando, *Architectural Acoustics* (AIP Press/Springer-Verlag, New York, 1998)]. As example for the control of $(te)_{min}$ due to performing style, effects of singing style, kind of vowel, relative pitch, vibrato extent, and intonation on the values of $(te)_{min}$ are demonstrated. In addition, the fine structure of the r -ACF is discussed with regard to the identification of vowels of singing voice. ^{a)}Now at 1-10-27 Yamanokami, Kumamoto, Japan.

11:00

2aAAb3. Acoustical qualification of Teatro Nuovo in Spoleto before refurbishing works. Alessandro Cocchi, Marco Cesare Consumi, and Ryota Shimokura (DIENCA, Eng. Faculty, Viale Risorgimento 2, 40136 Bologna, Italy)

To qualify the acoustical quality of an opera house two different approaches are now available: one is based on responses of qualified listeners (subjective judgments) compared with objective values of selected parameters, the other on comparison tests conducted in suited rooms and on a model of the auditory brain system (preference). In the occasion of the refurbishment of an opera house known for the Two Worlds Festival edited yearly by the Italian Composer G. C. Menotti, a large number of measurements were taken with different techniques, so it is possible to compare the different methods and also the results with some geometrical criterion, based on the most simple rules of musical harmony, now neglected as our attention is attracted to computer simulations, computer aided measurement techniques and similar modern methods. From this work some link between well known acoustical parameters (not known at the time when architects sketched the shape of ancient opera houses) and geometrical criteria (well known at the time when ancient opera houses were built) will be shown.

11:20

2aAAb4. Proposal of a new plane shape of an opera house—optimized by genetic algorithms. Takuya Hotehama, Yoichi Ando, Akinori Tani, and Hiroshi Kawamura (Grad. School of Sci. and Technol., Kobe Univ., Rokkodai, Nada, Kobe 657-8501, Japan, 002d885n@y02.kobe-u.ac.jp)

The horseshoe-shaped theater has been the main shape from historical circumstances. However, from acoustical points of view, the rationality of the peculiar plane shape is not yet verified more than historical refinement. In this study, in order to make the theater shape more acoustically excellent, optimization for temporal and spatial factors in the theory of the subjective preference was made using genetic algorithms (GAs) by operating the positions of side walls. Results reconfirm that the plane shape of the optimized theater is a leaf shape, which has been verified to be acoustically rational in a concert hall. And, further possible shapes are also offered.

Contributed Paper

11:40

2aAAb5. The control of balance between singer on the stage and orchestra in the pit by means of virtual opera house models. Linda Parati, Roberto Pompili, and Nicola Prodi (Dept. of Eng., Univ. of Ferrara, Via Saragat 1, 44100 Ferrara, Italy, lparati@ing.unife.it)

In this study some of the architectural elements which can influence the balance between the singer on the stage and the orchestra in the pit, in respect to the design constraints, have been investigated. By means of

room simulations, modifications in the type of materials, dimensions, and shapes of the pit, and modifications of the proscenium dimension and stage slope were applied. Some of the architectural elements controlling the balance have been identified and optimized. Simulations were performed with omnidirectional sources in the pit, despite the directivity of the instruments, and directional source on the stage. The effects of the applied changes on the balance between singer, conductor, and orchestra have been analyzed too.

Session 2aAAc**Architectural Acoustics: Theaters for Drama Performance—Another Two Decades (1984–2004)
(Poster Session)**

Robin Glosemeyer, Chair

*Jaffe Holden Acoustics, Inc., 501 Santa Monica Boulevard, Santa Monica, California 90401****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.

2aAAc1. Albert Ivar Goodman Theatre, The Goodman Theatre. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc7. Owen Bruner Goodman Theatre, The Goodman Theatre. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc2. Bingham Theatre, Actors Theatre of Louisville. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc8. The O'Reilly Theater, Pittsburgh Public Theater. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc3. Center Theater, California Center for the Arts. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc9. The Allen Theatre, ACT Theatre. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc4. Chicago Shakespeare Theater. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc10. The Falls Theatre, ACT Theatre. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc5. Lookingglass Theatre. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc11. Fichandler Stage, Arena Stage. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc6. Steppenwolf Theatre. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc12. The Cradle, Arena Stage. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc13. Sidney Harman Hall, Harman Center for the Arts. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc14. Second Theatre, Children's Theatre Company. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc15. Thrust Stage, Guthrie on the River. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc16. Proscenium Stage, Guthrie on the River. Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301, rick@talaske.com)

2aAAc17. Rodgers Victory Theatre. Steven Thorburn (P.O. Box 20399, Castro Valley, CA 94546, sjt@ta-inc.com)

2aAAc18. Drama Theater of the New National Theater (NNT), Tokyo, Japan. Takayuki Hidaka (Takenaka R&D, 1-5-1, Otsuka, Inzai, Chiba 270-1395, Japan) and Leo L. Beranek (Cambridge, MA 02138-5755)

2aAAc19. Roger S. Berlind Theatre, McCarter Theatre Center, Princeton, NJ. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

2aAAc20. New Amsterdam Theater. Robin Glosemeyer (Jaffe Holden Acoustics, Inc., 501 Santa Monica Blvd., Ste. 606, Santa Monica, CA 90401, rglosemeyer@jhacoustics.com)

2aAAc21. Purnell Center for the Arts, Carnegie Mellon University. Robin Glosemeyer (Jaffe Holden Acoustics, Inc., 501 Santa Monica Blvd., Ste. 606, Santa Monica, CA 90401, rglosemeyer@jhacoustics.com)

2aAAc22. New Second Stage Theatre. Robin Glosemeyer (Jaffe Holden Acoustics, Inc., 501 Santa Monica Blvd., Ste. 606, Santa Monica, CA 90401, rglosemeyer@jhacoustics.com)

2aAAc23. Biltmore Theater. Robin Glosemeyer (Jaffe Holden Acoustics, Inc., 501 Santa Monica Blvd., Ste. 606, Santa Monica, CA 90401, rglosemeyer@jhacoustics.com)

2aAAc24. Sarofim Hall, The Hobby Center for the Performing Arts. Robin Glosemeyer (Jaffe Holden Acoustics, Inc., 501 Santa Monica Blvd., Ste. 606, Santa Monica, CA 90401, rglosemeyer@jhacoustics.com)

2aAAc25. Yale Drama Theater. Robin Glosemeyer (Jaffe Holden Acoustics, Inc., 501 Santa Monica Blvd., Ste. 606, Santa Monica, CA 90401, rglosemeyer@jhacoustics.com)

2aAAc26. Majestic Hall, Brooklyn Academy of Music. Robin Glosemeyer (Jaffe Holden Acoustics, Inc., 501 Santa Monica Blvd., Ste. 606, Santa Monica, CA 90401, rglosemeyer@jhacoustics.com)

2aAAc27. Experimental theater of the New National Theater Tokyo, Japan. Takayuki Hidaka (Takenaka R&D, 1-5-1, Otsuka Inzai, Chiba 270-1395, Japan, hidaka.takayuki@takenaka.co.jp) and Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138-5755)

2aAAc28. Kokos, The Theater Academy of Finland. Henrik Moller and Tarja Lahti (Akukon Oy Consulting Engineers, Kometintie 4 A, FIN-00380 Helsinki, Finland, henrick.moller@akukon.fi)

2aAAc29. Oulu City Theater. Henrik Moller and Anssi Ruusuvoori (Akukon Oy Consulting Engineers, Kometintie 4 A, FIN-00380 Helsinki, Finland, henrik.moller@akukon.fi)

2aAAc30. Neal-Marshall Education Center, Indiana University. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

2aAac31. Cultural Life Center, Roberts Wesleyan College. R. Kring Herbert (Ostergaard Acoustical Assoc., 200 Executive Dr., West Orange, NJ 07052, kherbert@acousticalconsultant.com)

2aAac40. Murat Theater and Temple. William J. Cavanaugh (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA, cstorch@cavtocci.com)

2aAac32. Acoustical design of the Bayside Performing Arts Center. Jason R. Duty and David R. Schwind (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, acoustics@cmsalter.com)

2aAac41. Pinkerton Academy Theater. Lincoln B. Berry (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA, cstorch@cavtocci.com)

2aAac33. Acoustical design of the Randall Jr. Museum Theater. David R. Schwind (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, acoustics@cmsalter.com)

2aAac42. Rhode Island College Music Building. Lincoln B. Berry (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA, cstorch@cavtocci.com)

2aAac34. Acoustical design of the Mendocino Community College Fine Arts Theatre. David R. Schwind (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, acoustics@cmsalter.com)

2aAac43. Neptune Theater. Peter Terroux (Atlantic Acoust. Assoc.), Gregory C. Tocci, and Christopher A. Storch (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA, cstorch@cavtocci.com)

2aAac35. Vacaville Community Theater acoustics. Kenneth W. Graven and David R. Schwind (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, acoustics@cmsalter.com)

2aAac44. Ship's Company Theater. Peter Terroux (Atlantic Acoust. Assoc., P.O. Box 2520, DEPS, Dartmouth, NS B2W 4A5, Canada)

2aAac36. Sound isolation design for REDCAT (Roy and Edna Disney/CalArts Theater). Cristina L. Miyar, Thomas A. Schindler, and David R. Schwind (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, acoustics@cmsalter.com)

2aAac45. Restoration of the Geary Theatre, San Francisco. Ewart A. Wetherill (Ewart A. Wetherill AIA, 28 Cove Rd., Alameda, CA 94502, redwetherill@sbcglobal.net)

2aAac37. Newburyport Firehouse Theater. William J. Cavanaugh and Timothy J. Foulkes (Cavanaugh Tocci Assoc., Inc., cstorch@cavtocci.com)

2aAac46. San Jose Repertory Theater acoustics. Philip N. Sanders and David R. Schwind (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, acoustics@cmsalter.com)

2aAac38. Capitol Center Theater. William J. Cavanaugh and K. Anthony Hoover (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA, cstorch@cavtocci.com)

2aAac47. Acoustical design for the Fairfield Center for Creative Arts. David R. Schwind (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, acoustics@cmsalter.com)

2aAac39. Westminster School Fine Arts Center. Timothy J. Foulkes (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA, cstorch@cavtocci.com)

2aAac48. Blodgett Hall Auditorium, Vassar College. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

2aAac49. Acoustical design of the ACM Theater ATM. Hideo Nakamura and Keiji Oguchi (Nagata Acoust., Inc., Hongo-Segawa Bldg. 3F, 2-35-10 Hongo Bunkyo-ku, Tokyo 113-0033, Japan, nakamura@nagata.co.jp)

2aAac50. The Theatre, The Carlsen Center, Johnson County Community College, Overland Park, Kansas. Robert C. Coffeen (School of Architecture and Urban Design, The Univ. of Kansas, Lawrence, KS 66045)

2aAac51. Saitama Arts Theater (Main Theater). Shinichi Sawara, Fukushi Kawakami (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan, sawara@beat.yamaha.co.jp), and Kiyoteru Ishii (Univ. of Tokyo, Tokyo, Japan)

2aAac52. Hakata-za. Takayuki Watanabe, Takashi Yamakawa, and Yasushi Shimizu (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan, watanabe@beat.yamaha.co.jp)

2aAac53. Nagakute Cultural Center Mori-no Hall. Masato Hata, Fukushi Kawakami (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan, hata@beat.yamaha.co.jp), Yuji Sonoda, and Hideki Tachibana (Inst. of Industrial Sci., Univ. of Tokyo)

2aAac54. Arkas SASEBO. Masahiro Ikeda, Shinji Kishinaga, and Fukushi Kawakami (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan, ikeda@beat.yamaha.co.jp)

2aAac55. Kani Public Arts Center. Masahiro Ikeda, Shinji Kishinaga, Fukushi Kawakami (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan, ikeda@beat.yamaha.co.jp), and Masahito Yasuoka (Tokyo Univ. of Sci., Tokyo, Japan)

2aAac56. Noto Engekido. Takashi Yamakawa, Masato Hata, and Fukushi Kawakami (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan)

2aAac57. National Theater Okinawa. Takashi Yamakawa, Fukushi Kawakami (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan), Yukiya Tokuriki (Takamatsu Architects and Assoc. Co. Ltd., Osaka, Japan), Kiyoshi Masuda (Taisei Corp., Yokohama, Japan), and Kiyoteru Ishii (Univ. of Tokyo, Tokyo, Japan)

2aAac58. Roschel Performing Arts Center, Franklin & Marshall College. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

2aAac59. Biwako Hall, Center for the Performing Arts, Shiga. Shinji Kishinaga, Rento Tanase, and Fukushi Kawakami (Yamaha Corp., Advance System Development Ctr., Hamamatsu, Japan, kishinaga@beat.yamaha.co.jp)

2aAac60. Nohgaku-do, Niigata City Performing Arts Center. Yasushi Shimizu, Yoshikazu Honji, and Fukushi Kawakami (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan, shimizu@beat.yamaha.co.jp)

2aAac61. Theater, Niigata City Performing Arts Center. Yasushi Shimizu, Yoshikazu Honji, and Fukushi Kawakami (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan, shimizu@beat.yamaha.co.jp)

2aAac62. Civic Theater in Shonandai Culture Center. Yasushi Shimizu, Shinjiro Yamashita, and Fukushi Kawakami (Yamaha Corp., Adv. System Development Ctr., Hamamatsu, Japan, shimizu@beat.yamaha.co.jp)

2aAac63. Act City Hamamatsu Main Hall. Takayuki Watanabe, Shinji Kishinaga, and Fukushi Kawakami (Yamaha Corp. Adv. System Development Ctr., Hamamatsu, Japan, watanabe@beat.yamaha.co.jp)

2aAac64. Cultural Arts Center, Baton Rouge, LA. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

2aAac65. Hasty Pudding Theatre, Harvard University. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

2aAAc66. The Swisher Theater. Gary Siebein, Hyun Paek, Stephen Skorski, Robert Lilkendey (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), and Mac McGowan (McGowan Sound Design, Jacksonville, FL)

2aAAc67. Gulfport Community Theater. Gary Siebein, Hyun Paek, Stephen Skorski (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL, gsiebein@siebeinacoustic.com), and Michael Ermann (Virginia Polytechnic and State Univ.)

2aAAc68. The Players' Theater. Gary Siebein, Stephen Skorski, Hyun Paek (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), and Martin Gold (School of Architecture, Univ. of Florida)

2aAAc69. Horace Mann. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, New York 10801, info@acousticdimensions.com)

2aAAc70. Illinois State University Theater. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, New York 10801, info@acousticdimensions.com)

2aAAc71. Berrie Center—Ramapo College. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

TUESDAY MORNING, 25 MAY 2004

IMPERIAL BALLROOM B, 9:00 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics: Animal Bioacoustics Posters

Whitlow W. L. Au, Chair

Hawaii Institute of Marine Biology, P.O. Box 1106, Kailua, Honolulu 96734

Contributed Papers

All posters will be on display from 9:00 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 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

2aAB1. Hearing in the North Atlantic right whale: Anatomical predictions. Susan E. Parks,^{a)} Darlene R. Ketten, Jennifer Trehey O'Malley, and Julie Arruda (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, sparks@whoi.edu)

Understanding the hearing abilities of large whales is important to determine the impacts of anthropogenic sources of sound. Right whales are not amenable to traditional physiological techniques to test hearing. Previous research on the hearing of marine mammals has shown that functional morphometric models are reliable estimators of hearing sensitivity in marine species. Morphometric analyses of 18 inner ears from 13 stranded right whales were used in the development of a preliminary model of the frequency range of hearing. All ears were scanned with computerized tomography (CT). Four ears showing the best preservation were processed into slides for measurements of the basilar membrane. Calculated basilar-membrane length averaged 55.7 mm (range 50.5–61.7 mm). The ganglion cell density/mm averaged 1842 ganglion cells/mm. Membrane length and ganglion cell density were used to predict a total ganglion cell count of approximately 102,500 ganglion cells for right whales. The thickness/width measurements of the basilar membrane from slides resulted in an estimated frequency range of approximately 10 Hz–22 kHz based on established marine mammal models. Additional

measurements from more specimens will be necessary to develop a more robust model of the right whale hearing range. ^{a)}Currently at Cornell University Bioacoustics Research Program.

2aAB2. Song variation and environmental auditory masking in the grasshopper sparrow. Bernard Lohr, Robert J. Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742, blohr@psyc.umd.edu), and Douglas E. Gill (Univ. of Maryland, College Park, MD 20742)

Some grassland bird species, in particular grasshopper sparrows (*Ammodramus savaannarum*), sing songs with especially high mean frequencies (7.0–8.0 kHz). Acoustic interference is one potential explanation for the evolution of high frequency vocalizations, particularly in open habitats. We tested predictions from a model of effective auditory communication distances to understand the potential effects of vocal production and environmental auditory masking on vocal behavior and territoriality. Variation in the spectral structure of songs and the size and shape of territories was measured for grasshopper sparrows in typical grassland habitats. Median territory areas were 1629 m² at a site in the center of the species range in Nebraska, and 1466 m² at our study site in Maryland,

with average territory diameters measuring 20.2 m. Species densities and sound pressure levels also were determined for stridulating insects and other noise sources in the habitat. Based on current models of effective communication distances, known noise levels, and information on hearing abilities, our results suggest that auditory sensitivity and environmental noise could be factors influencing the mean frequency and spatial dynamics of territorial behavior in grassland birds. [Work supported by NIH and the CRFRC.]

2aAB3. Reduced temporal integration in Belgian Waterslager canaries. Amanda M. Lauer and Robert J. Dooling (Psych. Dept. and Ctr. for Comparative and Evolutionary Biol. of Hearing, Univ. of Maryland, College Park, MD 20743)

Belgian Waterslager canaries (BWS) are bred for their low-frequency song and have been shown to have hair cell abnormalities in the inner ear that result in elevated thresholds at higher frequencies. Previous results show that resolution of temporal fine structure, or the ability to resolve rapidly occurring changes in complex sounds, is enhanced in BWS canaries. In a continuing effort to assess the effects of the BWS inner ear pathology on hearing, we here investigate the ability to integrate acoustic information over longer periods of time. Absolute thresholds for 1.0-, 2.0-, and 4.0-kHz pure tones were measured in BWS and normal-hearing non-BWS canary strains for durations ranging from 5 to 480 ms using operant conditioning methods. NonBWS canaries showed a decrease in threshold of approximately 10–15 dB with increasing tone duration for all frequencies. In contrast, BWS showed almost no change in threshold across the range of durations tested for all frequencies. The reduced temporal integration in BWS canaries with hair cell abnormalities parallels similar findings in humans with cochlear damage. [Work supported by NIH DC01372 to RJD and DC05450 to AML.]

2aAB4. Changing the average frequency of contact calls is associated with changes in other acoustic parameters in the budgerigar (*Melopsittacus undulatus*). Michael Osmanski and Robert Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742)

The most-often produced vocalization of the budgerigar, a small parrot native to Australia, is the short (100–150 ms) frequency-modulated contact call. These calls play a role in maintaining flock dynamics and are believed to act as vocal signatures in these birds. Previous findings in our lab have shown that budgerigars can control the intensity of their vocal behavior and exhibit a robust Lombard effect (Manabe *et al.*, 1998). Recently, we have shown that there is a high degree of stereotypy in contact calls across a number of acoustic parameters (Osmanski and Dooling, 2004). Questions arise concerning the limits of plasticity in these calls and the relation or interdependence among the various parameters. As a first approach to answering these questions, four budgerigars were trained using operant conditioning methods to change the average peak frequency of

their contact calls (both upward and downward in frequency) to obtain access to a food reward. Results show that these birds can both increase and decrease the average frequency of their contact calls. Such changes are associated with modifications in a number of other acoustic parameters, suggesting constraints on vocal plasticity. [Work supported by NIH DC-00198 to RJD and NIDCD Training Grant DC-00046.]

2aAB5. An audiometric comparison of primate audiograms. Mark N. Coleman (Dept. of Anthropology, Stony Brook Univ., Stony Brook, NY 11794)

Audiogram data for 18 species of primates were collected from the literature and analyzed by measuring 13 audiometric variables: frequency and threshold of the primary peak, frequency and threshold of the secondary peak, frequency and threshold of the notch between peaks, low-frequency cutoff, high-frequency cutoff, total area of the audible field, low area, middle area, high area, and total audible range in octaves. All areal measurements were made using IGOR PRO 4.04 wave measurement software. Platyrrhines were found to have significantly better low-frequency sensitivity than like-sized lorisooids with an average of 15-dB difference between the means for the two groups. This difference remains significant even when interindividual variation is considered. *Callithrix jacchus* and *Erythrocebus patas* have unusual hearing patterns for primates of their size with marmosets showing a reduction in high-frequency sensitivity, while *patas* monkeys show a reduction in low-frequency sensitivity. It was also noted that chimps have a notch in sensitivity that falls within the range of greatest sensitivity for humans. These findings are discussed in relation to the morphological adaptations that appear to influence these hearing patterns and the evolutionary significance of such patterns for group communication and predator–prey interactions.

2aAB6. Influence of low-frequency vibration on changes of biochemical parameters of living rats. Cezary Kasprzak, Zbigniew Damijan, and Ryszard Panuszka (Univ. of Sci. and Technol., Lab. of Structural Acoust. and Biomed. Eng., Al. Mickiewicza 30, 30-059 Krakow, Poland, cekasp@agh.edu.pl)

The aim of the research was to investigate how some selected biochemical parameters of living rats depend on exposure of low-frequency vibrations. Experiments were run on 30 Wistar rats randomly segregated into three groups: (I) 20 days old (before puberty), (II) 70th day after; (III) control group. The exposure was repeated seven times, for 3 h, at the same time of day. Vibrations applied during the first tests of the experiment had acceleration 1.22 m/s² and frequency 20 Hz. At the 135th day the rats' bones were a subject of morphometric/biochemical examination. The results of biochemical tests proved decrease in LDL and HDL cholesterol levels for exposed rats as well as the Ca contents in blood plasma. There was evident increasing of Ca in blood plasma in exposed rats for frequency of exposition.

Session 2aAO

Acoustical Oceanography: Ocean Acoustics of Earthquakes

Ralph A. Stephen, Cochair

Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543-1542

Robert I. Odom, Cochair

*Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105-6698***Chair's Introduction—8:00*****Invited Papers*****8:05****2aAO1. Characteristics of *T* phases from earthquake sources, or how earthquakes “talk” into the ocean.** Emile A. Okal (Dept. of Geol. Sci., Northwestern Univ., Evanston, IL 60208, emile@earth.nwu.edu)

We review the mechanisms by which earthquake sources within the solid Earth generate acoustic energy into the SOFAR channel. We examine individual cases involving both the relatively inefficient mechanism of “downslope conversion,” leading to *T* phases with low amplitudes and long durations, and efficient conversion at steep interfaces allowing direct penetration of the SOFAR, larger amplitudes, and shorter durations. We derive several ways to quantify *T* waves generated by an earthquake source. To avoid the saturation of their amplitude as source size grows, we use the concept of a *T*-phase energy flux (TPEF), which mirrors the radiated energy measured on conventional seismic body waves. TPEF is scaled to the seismic moment of the earthquake to define a parameter, γ , characterizing the earthquake's efficiency for *T*-phase generation. In another approach, we compare the amplitude of *T*-phase envelopes to the duration of their signal. The resulting discriminant has been proposed to identify earthquakes from explosions. We show that it varies for several categories of nonstandard earthquakes, notably those occurring in volcanic islands and events with slow rupture giving rise to large tsunamis. Finally, we discuss a few examples of *T* waves from underwater landslides.

8:25**2aAO2. Wave-theory modeling of oceanic *T*-phase coupling at continental margins and seamounts.** Henrik Schmidt, Arthur B. Baggeroer (Dept. of Ocean Eng., MIT, Cambridge, MA 02139), and Brian J. Sperry (Sci. Application Intl. Corp., McLean, VA 22102)

The role of seismo-acoustic seabed scattering as a mechanism for coupling of seismic energy into oceanic teleseismic waves or *T*-phases is investigated using a new versatile modeling capability for seismo-acoustic propagation in laterally inhomogeneous or range-dependent ocean waveguides. The Virtual Source Approach (VISA) uses a local Rayleigh–Kirchhoff approximation to handle the transmission and reflection of plane waves at the vertical interfaces separating horizontally ocean stratified sectors. Combined with the wavenumber integration approach which inherently computes the plane-wave decomposition of the seismo-acoustic field in stratified fluid-elastic waveguides, this approach provides a robust approximation to the seismo-acoustic coupling phenomena in shallow and deep ocean waveguides. The VISA approach has been implemented in the OASES seismo-acoustic modeling framework and used to investigate the role of seismo-acoustic conversion and scattering by seabed topography and roughness in generating oceanic *T*-phases at continental margins and seamounts. It is demonstrated that the excitation of the oceanic *T*-phases can be explained by the coupling of crustal shear body-waves into seismic interface waves, or seabed Scholte waves, which then subsequently scatter into the waterborne modal spectrum. This wavenumber conversion mechanism implies that the excitation of the *T*-phases will be significantly stronger by earthquakes producing crustal SV-waves than those producing predominantly P-waves. This in turn suggests that earthquakes associated with dip-slip failure modes excite significantly stronger *T*-phases than buried explosive sources.

8:45**2aAO3. Computation of *T*-phase seismograms using stairsteps to model seafloor bathymetry changes.** Catherine de Groot-Hedlin (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093, chedlin@ucsd.edu)

Stair-step boundaries are employed to represent range-dependent seafloor bathymetry in many types of numerical modeling methods. However, if an overly coarse discretization is used to simulate smoothly varying bathymetry, the acoustic solution is degraded in a way that simulates scattering. Geometrical optics approximations can be used to derive discretization criteria for simulating a smoothly sloping interface for the case of a source embedded in either an acoustic or an elastic seafloor, and applied to modeling *T* phases. A finite difference time-domain modeling approach is used to synthesize *T* phases for both smoothly sloping and rough seafloor boundaries. It is shown that scattering at a rough seafloor boundary yields ocean-borne acoustic phases with velocities near those of observed *T* phase, while smooth seafloor models yield *T* phases with slower horizontal velocities. The long duration of the computed *T* phases for both the rough acoustic and elastic models is consistent with energy being scattered into the sound channel both as it transits the ocean/crust boundary, as well as at several subsequent seafloor reflections. However, comparison between the elastic and acoustic modeling solutions indicates that the *T*-phase wavetrain duration decreases with decreasing impedance contrast between the ocean and seafloor.

2aAO4. The use of regional hydrophone datasets for statistical studies of mid-ocean ridge earthquake populations. DelWayne R. Bohnenstiehl (Lamont-Doherty Earth Observatory of Columbia Univ., P.O. Box 1000, Palisades, NY 10964)

Seismically generated tertiary (T-) waves can be used to detect and locate the shallow hypocenter earthquakes associated with mid-ocean ridge spreading centers and oceanic transforms. During the last several years, regional seismic catalogs, which contain estimates of T-wave source regions, origin times and acoustic source levels, have been generated from continuous hydroacoustic monitoring along the northern Mid-Atlantic Ridge and equatorial East Pacific Rise. As the mechanics of T-wave generation remain poorly understood, critics of these datasets have questioned the correlation between T-wave source regions and seismic epicenters, as well as the usefulness of acoustic source level as a proxy for earthquake size. Recent analyses, however, suggest that these data provide a reasonably complete record of seismicity within the mid-ocean ridge environment, yielding significant improvements in location accuracy and catalog completeness level, relative to global seismic catalogs. Most importantly, it appears that many fundamental and well-established properties of seismic distributions can be studied using T-wave catalogs, including the modified Omori decay law of aftershocks, a power-law size frequency distribution (Gutenberg-Richter law), a fractal time-clustering behavior ($1/f$ noise) and fractal spatial distribution of epicenters. These observations show that T-wave derived earthquake catalogs can be used for quantitative seismo-tectonic studies in the ridge setting.

2aAO5. Hydroacoustic observations of sub-Indian ocean earthquakes using arrays of the International Monitoring System. Jay J. Pulli and Zachary M. Upton (BBN Technologies, 1300 N. 17th St., Ste. 400, Arlington, VA 22209, jpulli@bbn.com)

Until recently, there have been few public hydroacoustic recordings of earthquakes beneath the Indian Ocean. Now, the International Monitoring System, which includes three Indian Ocean station sites, is providing excellent data for study. These stations consist of three-element horizontal hydrophone arrays floated to the middle of the sound channel axis, and can record low-level signals from distant sources. System bandwidth is approximately 2–100 Hz. We have accumulated a waveform database of 150 sub-Indian Ocean earthquakes and surrounding continental events recorded by this system. These data allow us to research *T*-wave generation and propagation. The recordings include seismic body-wave conversions beneath the arrays, *T* waves, and long-range reflections. Recordings of main shock/aftershock series have allowed us to compute scaling relationships for *T*-wave amplitudes versus magnitude and examine *T*-wave excitation versus source mechanism. When combined with distance corrections, we can scale *T* waves recorded at a given station to a fixed magnitude and distance. Plotted versus azimuth around the station, these scaled *T*-wave amplitudes provide insight into hydroacoustic blockage, which is essential for understanding detection capability. Seismic body-wave conversions beneath the arrays can be timed to 1 second or better, and provide important location constraints for subocean events occurring far from continental stations.

2aAO6. Directionality of acoustic *T*-phase signals from shallow submarine earthquakes. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada, chapman@uvic.ca)

Acoustic transients radiated from shallow undersea earthquakes were studied in an experiment carried out using a towed horizontal line array operating in the South Fiji Basin. The transient signals consisted of *P*, *S*, and *T* phases, with the *T*-phase signal from each earthquake lasting for about 5 min. During this period, the directionality of the *T*-phase signal was determined by processing the array data with a conventional beamformer. The weakest part of the signal arrived first on a direct bearing between the earthquake source and the array. However, subsequent stronger components of the *T* phase arrived from different directions farther south of the source, in a region where a number of seamounts rose within the sound channel. A simple model based on ray-path travel times for *p*-wave travel in the earth and in the water suggests that the later components of the *T*-phase signal are radiated into the water by downslope propagation from the seamounts and ridges. The initial weaker components may be scattered into the sound channel by leakage from the *P* and *S* phases relatively closer to the array.

10:05–10:20 Break

Contributed Papers

10:20

2aAO7. Modal scattering and T-waves: Sediment amplification and source effects. Robert I. Odom (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105) and Darin J. Soukup (Univ. of Washington, Seattle, WA 98195)

If the Earth were a plane-layered semi-infinite half-space or a radially symmetric sphere oceanic, T-waves could not exist. This is apparent because the source depth of T-wave producing earthquakes is greater than the depth at which the low order modes comprising the T-waves have any significant amplitude. Bottom roughness provides a mechanism by which energy from high order source modes can be scattered into the lower order modes. The efficiency of this scattering is improved when there is a layer of sediments overlying the higher speed upper ocean crust. Some of the source modes develop a large anti-node, reminiscent of a Scholte wave, at the water sediment interface. Bottom roughness serves as a sheet of secondary sources placed directly on the anti-node of these modes, and con-

tributes a significant amount of energy to the T-waves. However, a significant fraction of the total T-wave energy is also provided by small scattered contributions from many higher modes of the continuum spectrum, which is modeled using the locked mode approximation. Source mechanism modeling of T-wave excitation shows that normal fault earthquakes are inefficient generators of T-waves.

10:35

2aAO8. Coupling of acoustic normal modes and seismo-acoustic surface waves in variable-depth shallow water. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA/Environ. Technol. Lab., Mail Code R/ET1, 325 Broadway, Boulder, CO 80305)

Marine sediments support Rayleigh-type surface waves which can propagate along the sea floor in arbitrarily shallow water and even on shore. At frequencies above a few Hertz, because of a strong attenuation of compression and especially shear waves in the sediments, the surface

waves can significantly contribute to the acoustic field far from the shore only through their coupling to volume waves in the water. We study theoretically excitation of acoustic normal modes by the seismo-acoustic surface waves in a shallow-water waveguide with a sloping bottom. The role of bottom stratification is assessed by comparing acoustic fields generated by the Rayleigh wave, which is supported by a homogeneous elastic bottom, and by the fundamental mode of surface waves, which are supported by the bottom with a linear profile of shear rigidity. It is found that the coupling primarily occurs in the vicinity of a modal cutoff. The first normal mode is the one most strongly excited by the surface waves. For realistic values of geoacoustic parameters, the energy conversion rate can reach tens of percent, making the surface-to-volume wave conversion an efficient mechanism of the oceanic waveguide excitation by acoustic sources on shore or in very shallow water.

10:50

2aAO9. Listening to distant earthquakes with hydrophones on autonomous oceanic floats. Frederik Simons, Guust Nolet (Princeton Geosciences, Guyot Hall, Princeton, NJ 08540, fjsimons@princeton.edu), and Jeff Babcock (UCSD, La Jolla, CA 92093)

We are mounting a hydrophone on autonomous floats capable of drifting below the sound channel and surfacing to communicate data by a satellite link. Using new, intelligent algorithms for the automatic identification and discrimination of seismic phases, we recognize teleseismic arrivals in the presence of local *P*, *S*, and *T* phases, ship and whale noise, and other contaminating factors such as airgun surveys. Our approach combines time-domain methods with spectrogram analysis and with wavelet methods. To maximize battery life, we optimize the efficiency of our algorithms and their numerical implementation. Our algorithms were tested on data from tethered hydrophones from two arrays anchored to the Mid-Atlantic Ridge and the East Pacific Rise. Our prototype device was successfully tested in a dive to 700 m off the coast of La Jolla. We acquired

a valuable 31-h data stream suitable to characterize the ambient noise and were able to identify a number of engineering problems with the hydrophone sensitivity. We expect detecting teleseisms with magnitudes superior to 6.3, and thus adding at least 200 high-quality recordings to the global catalog over the life span of a single instrument—well worth the \$15 000 manufacturing price and negligible deployment costs on ships of opportunity.

11:05

2aAO10. Hydroacoustic events located near the Atlantis (30N) and Kane (2330N) transform faults on the MAR. Clare M. Williams, Ralph A. Stephen, and Deborah K. Smith (WHOI, 360 Woods Hole Rd. (MS 24), Woods Hole, MA 02543-1542, rstephen@whoi.edu)

Hydroacoustic arrivals detected by a hydrophone array on the Northern Mid-Atlantic Ridge (MAR) are used to better understand *T* phases from oceanic crust earthquakes. *T*-phase events are selected from two study areas: the inside corner and ridge transform intersection at the eastern ends of the Kane (2330N) and Atlantis (30N) transform faults. Both are regions of high relief (5000 m) and events are located throughout the massif and transform valleys. We investigate the spatial distribution of *T*-phase locations as a function of water depth. We examine the characteristics of *T* phases as a function of both distance from the event to each hydrophone and event location water depth. Finally, we use a ray-trace model to test for bathymetric blockage along the propagation path of the *T* phase. We observe that acoustic magnitudes of *T* phases show no dependence on water depth of event location. This is opposite to predictions from current *T*-phase generation models which show water depth dependence. There may be a correlation between acoustic magnitude and propagation path topography, but results are sensitive to the sound velocity profile used in the model. Our results underscore the complexity of *T*-phase generation and propagation.

TUESDAY MORNING, 25 MAY 2004

NEW YORK BALLROOM A, 7:55 A.M. TO 12:00 NOON

Session 2aBB

Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: High Intensity Focused Ultrasound I

Cheri Deng, Cochair

Case Western Reserve University, 10900 Euclid Avenue, Cleveland, Ohio 44106

Shahram Vaezy, Cochair

University of Washington, HSB Box 356540, Seattle, Washington 98195-6540

Chair's Introduction—7:55

Invited Papers

8:00

2aBB1. The feasibility of noninvasive image-guided treatments of brain disorders by focused ultrasound. Kullervo Hynynen, Gregory Clement, Nathan McDannold, Natalia Vykhodtseva (Brigham and Women's Hospital and Harvard Med. School, 75 Francis St., Boston, MA 02115, kullervo@bwh.harvard.edu), P. Jason White, Ferenc A. Jolesz, and Nikolai A. Sheikov (Brigham and Women's Hospital and Harvard Med. School, Boston, MA 02115)

Brain disorders, such as tumors, functional problems, etc., are difficult to treat and the invasive interventions often disturb surrounding brain tissue, resulting in complications. In addition, the delivery of therapeutic agents to the brain via the blood supply is often impossible because the blood brain barrier protects the brain tissue from foreign molecules. Our hypothesis has been that transcranial therapeutic ultrasound exposures can be delivered with an optimized phased-array system. We have demonstrated that highly focused therapeutic ultrasound beams can be accurately delivered through an intact human skull noninvasively. Furthermore, we demonstrated using *ex vivo* human skulls that we can use CT-derived information to predict the phase shifts required for correcting

the wave distortion. We have also developed a method to focally disrupt the blood brain barrier without damaging the neurons in the targeted tissue volume. This may allow delivery of therapeutic or diagnostic agents into image-specified locations. Successful transcranial delivery of ultrasound in a clinical setting may have a major impact on the treatment of brain disorders in the future.

8:20

2aBB2. Design and testing of an ultrasonic system with integrated diagnostic and therapy modes. Frederic L. Lizzi (Riverside Res. Inst., 156 William St., New York, NY 10038-2609)

Our laboratories developed and tested a system integrating high-intensity focused ultrasound (HIFU) with diagnostic ultrasound procedures for aiming, delivering, and monitoring treatment exposures. The system employs 5-MHz, 5-element spherical-cap HIFU arrays with annular electrodes (for variable focusing) or strip electrodes (for inducing broad lesions of controlled width). Central apertures in these arrays house a coaxial diagnostic phased array (48 elements), interfaced with a custom digital system for visualization and RF echo acquisition. Several operational modes were investigated. A preexposure mode permits the HIFU focal point to be aimed at the desired location in B-mode images. It also assesses harmonic generation in the HIFU beam, using the diagnostic array to receive and spectrally analyze echoes from a short HIFU pulse aimed at targeted tissues. Radiation-force elastography uses the diagnostic array to monitor tissue displacements induced by brief exposures from the HIFU transducer; pre- and posttherapy data are compared to delineate mechanical alterations in lesioned volumes. Acquired pre- and post-RF data are also spectrally analyzed to map changes in scattering and attenuation indicative of lesion production. These modes have been successfully tested under *in vitro* conditions and are now entering use in animal studies aimed at treating cancer and cardiac diseases.

8:40

2aBB3. The physical acoustics of HIFU exposure: A suite of *in vitro* studies. Ronald A. Roy, R. Glynn Holt, and Robin O. Cleveland (Dept. of Aersp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, ronroy@bu.edu)

Exposing tissue and blood media to high-intensity focused ultrasound leads to a broad range of physical and biological effects. In addition to tissue heating from thermo-viscous absorption, one encounters nonlinear and second order phenomena such as wave-front shocking, radiation stress, acoustic streaming, bubble nucleation and cavitation. All of these processes impact one's ability to effect controlled tissue heating. This paper reviews a suite of studies under way at Boston University that focus on developing a quantitative understanding of the physical acoustics of high-intensity focused ultrasound. We describe research into HIFU heating and both uniform and flow-through tissue phantom and address the potentially critical roles played by bubbles and acoustic cavitation. [Work supported by the U.S. Army and the Center for Subsurface Sensing and Imaging Systems via NSF ERC Award No. EEC-9986821.]

9:00

2aBB4. Approaches to overcome current limitations of HIFU treatment. Shin-ichiro Umemura, Ken-ichi Kawabata, Kazuaki Sasaki, Takashi Azuma (Central Res. Lab., Hitachi Ltd., Kokubunji, Tokyo 185-8601, Japan, sumemura@crl.hitachi.co.jp), Kazunari Ishida, Jun Kubota (Hitachi Medical Corp., Chiba 277-0804, Japan), Mitsuyoshi Ichihara, and Takashi Okai (Showa Univ. School of Medicine, Tokyo 142-8666, Japan)

Noninvasive therapy with HIFU has been successfully applied to transrectal treatment of prostate cancer as well as benign prostate hyperplasia. However, there are two major technical reasons why its clinical application to other organs is currently limited: (1) low throughput of treatment and (2) lack of penetration to deep tissues. To multiply the throughput, a split-focus technique, in which the focal spot is enlarged primarily in the lateral direction, was developed. An electronically variable focus array transducer was also developed to enhance the throughput. An approach to treat a large volume of uterus myoma by coagulating its feeding arteries has been studied. The tissue volume to be coagulated can be thereby reduced by orders of magnitude. The penetration and throughput can potentially be improved at the same time by delivering a microbubble agent to the target tissue. It was theoretically predicted that a microbubble agent could multiply the ultrasonic tissue absorption. The effectiveness of this approach was confirmed in animal experiments using Optison. Real-time monitoring of tissue coagulation during HIFU exposure also can enhance the throughput through preventing excess deposition of ultrasonic energy. Monitoring coagulation by imaging local displacement in tissue with ultrasound will be discussed as well.

9:20

2aBB5. Thermal ablation by interstitial ultrasound. Cyril Lafon, David Melodelima, Jean-Yves Chapelon, and Dominique Cathignol (INSERM U556, 151 Cours Albert Thomas, 69003 Lyon, France)

Interstitial ultrasound applicators have been proposed for treating deep-seated tumors that are unreachable with extracorporeal high intensity focused ultrasound. The technique consists of bringing the ultrasound source as close as possible to the target in order to minimize the effects of attenuation and phase aberration along the ultrasound pathway. We have designed interstitial applicators with flat ultrasound transducers that operate at a frequency between 3.5 and 20 MHz. Nonfocused transducers require working at high frequencies to obtain coagulation in a short period of time. Flat transducers were chosen because the pressure drop in the near field is only due to absorption, and extended therapeutic depth can be achieved. A rotation of the transducer results in a cylindrical or sector-based volume of necrosis. To avoid these mechanical displacements, a cylindrical array has been manufactured; it allows for electronic rotation of a plane wave. Imaging modalities like endoscopic ultrasound and MRI were tested for guiding our treatments. Clinical trials are now being performed for the thermal ablation of digestive tumors and preliminary results are extremely promising.

9:40

2aBB6. Comparison between coupled KZK-BHTE numerical simulations and scanned HIFU exposures in excised bovine liver.

Marilee A. Andrew, Andrew A. Brayman, Peter J. Kaczkowski, and Steven G. Kargl (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698)

The use of moving high intensity focused ultrasound (HIFU) treatment protocols is of interest in achieving efficient formation of large-volume lesions in tissue. However, potentially unwanted thermal effects, such as pre-focal heating, should be considered. A KZK acoustic model coupled with the BioHeat Transfer Equation has been extended to simulate multiple, moving scans in tissue. Simulation results are compared with experimental data collected over a range of exposure regimes for linear and concentric circular scans with a 3.5-MHz single-element transducer in *ex vivo* bovine liver. Of particular interest are investigating pre-focal thermal buildup and ablating the central core of a circular pattern through conductive heating, that is without direct HIFU exposure. Qualitative agreement is observed between experimental and simulated data; limits of the predictive capability of the model in cavitation regimes will be discussed. [Support provided by the U.S. Army Medical Research Acquisition Activity through The University of Mississippi under terms of Agreement No. DAMD17-02-2-0014. The opinions expressed herein are those of the author(s) and do not necessarily reflect the views of U.S. Army Medical Research Acquisition Activity or The University of Mississippi.]

9:55

2aBB7. Cavitation detection during and following HIFU exposure *in vitro*.

Constantin-C. Coussios (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, UK, constantin.coussios@eng.ox.ac.uk), Caleb H. Farny, Charles R. Thomas, Robin O. Cleveland, R. Glynn Holt, and Ronald A. Roy (Boston Univ., Boston, MA 02215)

Is the existence of a sustained hyperechogenic region on B-scan images following HIFU treatment a necessary and sufficient condition for cavitation to have occurred during HIFU exposure? Three means of cavitation monitoring were used synchronously, before, during, and after continuous-wave HIFU exposure of an agar-graphite tissue phantom. A 1.1-MHz HIFU transducer was confocally aligned with a 15-MHz passive cavitation detection (PCD) transducer and a 5-MHz scan head. The HIFU pressure amplitude was increased in steps of 0.26 MPa every 5 s. A peak detector recorded the peak PCD signal level and a dynamic signal analyzer monitored broadband noise emissions (5–10 MHz). The sudden onset of a PCD output signal occurred at a peak-negative focal pressure of 1.25 MPa; no post-HIFU hyperechogenic region was visible on the B-scan images. A hyperechogenic region did eventually appear, but only for focal pressures in excess of 1.8 MPa. Inertial cavitation can therefore occur during HIFU exposure in the absence of a post-HIFU hyperechogenic region. The focal pressure for which such a region is observed can be as much as 50% higher than the threshold pressure for inertial cavitation inception. [Work supported by the ASA and the NSF Center for Subsurface Sensing and Imaging Systems.]

10:10–10:30 Break

10:30

2aBB8. Fluorescent real-time monitoring of HIFU cardiac focal ablation.

Fujian Qu, Vladimir Nikolski, Igor Efimov, and Cheri Deng (Dept. of Biomed. Eng., Case Western Reserve Univ., 10900 Euclid Ave., Cleveland, OH 44106-7207, cxd54@cwru.edu)

To study HIFU cardiac ablation, fluorescent imaging was used to monitor in real time the electrophysiology changes of cardiac tissues during focal HIFU ablation. We applied HIFU ablation of AV nodal and ventricular preparations of Langendorff-perfused rabbit heart while monitoring electrical activity in real-time. HIFU energy was applied to ablate the AV node and ventricular tissue of Langendorff-perfused rabbit hearts while

monitoring electrical activity in real-time with fluorescent voltage-sensitive dye imaging and surface electrodes. HIFU was generated using a spherical piezoelectric ceramics transducer (diameter 42 mm, F-number 1.2) at 4.23 MHz. When HIFU was applied to ventricular epicardium fluorescent imaging it revealed gradual reduction of the plateau phase and amplitude of the action potential. Subsequently conduction block and cell death were observed at the site of ablation. In our study HIFU produced focal lesions of 0.2–0.8 mm for 10–60-s applications. When HIFU was applied to the AV node, fluorescent imaging and electrograms revealed the development of the AV block.

10:45

2aBB9. Feasibility of *in vivo* cardiac HIFU ablation.

Kana Fujikura, Ryo Otsuka, Shunichi Homma (Columbia Presbyterian Med. Ctr. PH3-133, 630 W. 168th St., New York, NY 10032, kf2113@columbia.edu), Robert Muratore, Jeffrey A. Ketterling, and Frederic L. Lizzi (Riverside Res. Inst., New York, NY 10038-2609)

The potential for cardiac applications of HIFU remains unexamined. In order to create reproducible lesions in a beating heart, it is necessary to maintain focusing at a certain position within moving myocardial tissue. One technique is to use multiple short HIFU exposures (0.2 s) and synchronize them with an EKG signal and respiration. The left ventricular free wall (LVFW) of calf hearts were cut into 4-cm cubes, degassed in phosphate buffer saline (PBS), and heated to 37 °C. An 80-mm-diam spherical-cap transducer with a focus of 90 mm was operated at a frequency of 4.67 MHz and a nominal focal point intensity of 26.9 kW/cm². The transducer was coupled to the LVFW using degassed PBS. First, the effect of pericardial fat, focal depth, and temperature on lesion size was individually evaluated. Then the 0.2-s HIFU exposure was applied 10 to 30 times at 4-s intervals. The same HIFU transducer was applied to an open-chest canine LVFW with the same triggering protocol. Dimensions of all lesions were measured by visual examination of the fresh, unstained tissue. A histopathological examination of the lesion was also performed. The *in vivo* lesions were created in similar size to those *in vitro*.

11:00

2aBB10. Experimental high-intensity focused ultrasound lesion formation in cardiac tissue.

Robert Muratore, Andrew Kalisz, Paul Lee, Frederic Lizzi (Riverside Res. Inst., 156 William St., New York, NY 10038-2609, muratore@rrinyc.org), Kana Fujikura, Ryo Otsuka, and Shunichi Homma (Columbia Univ. College of Physicians and Surgeons, New York, NY 10032)

High-intensity focused ultrasound (HIFU) (4.5–7.5 MHz) was used to form lesions in cardiac tissue, with an ultimate objective of treating conditions such as hypertrophic cardiomyopathy and ventricular tachycardia. Ultrasound attenuation coefficients were experimentally determined *in vitro* for calf myocardial tissue, both muscle and pericardial fat. These coefficients were employed in computational models of linear beam propagation, tissue heating profiles and thermal lesion formation for a variety of focused transducers. Modeling was performed for continuous and pulsed exposures. These models suggested initial power levels and exposure durations for *in vitro* experiments on calf ventricles and septa and *ex vivo* experiments on canine whole hearts. Repeatability of lesion size and placement was studied as power and exposure parameters varied around the initial values. With these experimental results, power and exposure parameters were selected to create lesions *in vivo* in canine ventricles and septa in open-chest, anesthetized dogs. Pulsed exposures were synchronized to cardiac and respiration cycles to ensure accurate placement of the lesions. These initial *in vivo* experiments showed that HIFU treatments in the beating heart are feasible; they also identified refinements that are now being implemented for better control of lesion size and placement. [Work supported by NCI and NHLBI Grant 5R01 CA84588.]

11:15

2aBB11. The relative effects of cavitation and nonlinear ultrasound propagation on HIFU lesion dynamics in a tissue phantom. Vera A. Khokhlova (Dept. of Acoust., Faculty of Phys., M. V. Lomonosov Moscow State Univ., Leninskie gory, Moscow 119992, Russia, vera@acs366.phys.msu.ru), Michael R. Bailey, Justin Reed, and Peter J. Kaczkowski (Univ. of Washington, Seattle, WA 98105)

The relative importance of the effects of acoustic nonlinearity and cavitation in HIFU lesion production is studied experimentally and theoretically in a polyacrylamide gel. A 2-MHz transducer of 40-mm diameter and 45-mm focal length was operated at different regimes of power, and in cw or duty-cycle regimes with equal mean intensity. Elevated static pressure was applied to suppress bubbles, increase boiling temperature, and thus to isolate the effect of acoustic nonlinearity in the enhancement of lesion production. Experimental data were compared with the results of simulations performed using a KZK acoustic model combined with the bioheat equation and thermal dose formulation. Boiling and the typical tadpole-shaped lesion shifting towards the transducer were observed under standard atmospheric pressure. No boiling was detected and a symmetric thermal lesion formed in the case of overpressure. A delay in lesion inception time was registered with overpressure, which was hypothesized to be due to suppressed microbubble dynamics. The effect of acoustic nonlinearity was revealed as a substantial decrease in the lesion inception time and an increase in the lesion size for high-amplitude waves under both standard and overpressure conditions. [Work supported by ONRIFO, NASA/NSBRI, NIH Fogarty, and CRDF grants.]

11:30

2aBB12. Effect of nonlinearity on lesion formation for high-intensity focused ultrasound (HIFU) exposures. Paul Lee, Frederic L. Lizzi, Jeffrey A. Ketterling (Riverside Res. Inst., 156 William St., New York, NY 10038, lee@rrinyc.org), and Christopher J. Vecchio (Spectrasonics Imaging, Inc., Wayne, PA 19087)

This study examined the effects of nonlinear propagation phenomena on two types of HIFU transducers (5 MHz) being used for thermal treatments of disease. The first transducer is a 5-element annular array. The second is a transducer with a 5-strip electrode; its multilobed focused beam is designed to efficiently produce broad, paddle-shaped lesions. The

beam patterns of these transducers were computed using a variety of excitation patterns for electronic focusing of the annular array and variation of lesion size for the strip-electrode transducer. A range of intensities was studied to determine how nonlinear propagation affects the beam shape, constituent frequency content, grating lobes, etc. These 3D computations used a finite-amplitude beam propagation model that combined the angular spectrum method and Burger's equation to compute the diffraction and nonlinear effects, respectively. Computed beam patterns were compared with hydrophone measurements for each transducer. The linear and nonlinear beam patterns were used to compute the absorbed thermal dose, and the bioheat equation was evaluated to calculate 3D temperature rises and geometry of induced lesions. Computed lesion sizes and shapes were compared to *in vitro* lesions created by each HIFU transducer. [Work supported by NCI and NHLBI Grant 5R01 CA84588.]

11:45

2aBB13. Experimental verification of enhancement of HIFU-induced heating of tissue mimicking phantoms due to acoustic nonlinearity. Tatiana V. Sinilo, Vera A. Khokhlova, and Oleg A. Sapozhnikov (Dept. of Phys., Moscow State Univ., Leninskie Gory, Moscow 119992, Russia, tanya@acs366.phys.msu.ru)

A possibility of enhancement of heat deposition of high-intensity focused ultrasound (HIFU) in tissue mimicking phantoms due to acoustic nonlinearity is experimentally studied. A gelatin sample of high concentration was used to model medium with acoustic properties similar to those of biological tissue. Several sets of pulse-periodic regimes with the same mean power but different pulse amplitudes and durations were used. In regimes with higher amplitudes (lower duty cycles), the waveform in the focal region was more strongly distorted because of acoustic nonlinearity. In the linear medium all the regimes of one set would give the same heating. Optical shadow pictures of the HIFU-heated region were taken in the regimes with different duty cycles but the same mean power. The heated region size increased for lower duty cycles, but the shadow shape remained regularly cigar-like in all the regimes, even when the ultrasound waveform was shocked. The transmitted signal measured by a hydrophone was stable. This showed that cavitation was not pronounced. Thermocouple measurements and theoretical modeling of gelatin temperature in the focal region of the ultrasound beam showed significant increase of HIFU heat deposition in the presence of shocks in the waveform. [Work supported by CRDF and RFBR.]

TUESDAY MORNING, 25 MAY 2004

CONFERENCE ROOM K, 8:00 TO 10:15 A.M.

Session 2aMUa

Musical Acoustics: Measurement and Analysis

James P. Cottingham, Chair
Department of Physics, Coe College, Cedar Rapids, Iowa 52402

Contributed Papers

8:00

2aMUa1. Resonant and nonresonant sound radiation from a guitar. Richard H. Lyon (RH Lyon Corp, 691 Concord Ave., Cambridge, MA 02138)

When forces are applied to a flexible structure, a worker in structural acoustics who is concerned with the radiated sound will normally be interested in how much of the sound is arising from the resonant response of

the structure compared to the nonresonant response. Lightweight panels below their critical frequency are likely to have a significant amount of sound arising from nonresonant response due to mass inertial response localized around the location of force application. Since a guitar top is such a structure, it is natural to inquire about the comparison between resonant and nonresonant radiation from a guitar over its normal range of frequencies. Measurements have been made on a student Yamaha and a concert quality Modranas/Connor guitar. The results of these measurements and their interpretation will be discussed.

8:15

2aMUa2. Noncontact modal analysis of a pipe organ reed using airborne ultrasound stimulated vibrometry. Thomas M. Huber (Dept. of Phys., Gustavus Adolphus College, 800 College Ave., Saint Peter, MN 56082, huber@gustavus.edu), Mostafa Fatemi, Randall R. Kinnick, and James F. Greenleaf (Mayo Clinic College of Medicine, Rochester, MN 55905)

The goal of this experiment was to excite and measure, in a noncontact manner, the vibrational modes of the reed from a reed organ pipe. To perform ultrasound stimulated excitation, two ultrasound beams in air of different frequencies were directed at the reed; the audio-range beat frequency between these ultrasound beams induced vibrations. The resulting vibrational deflection shapes were measured with a scanning vibrometer. The modes of any relatively small object can be studied in air using this technique. For a 36 mm by 7 mm clamped brass reed cantilever, displacements and velocities of 5 μ and 4 mm/s could be imparted at the fundamental frequency of 145 Hz. Using the same ultrasound transducer, excitation across the entire range of audio frequencies was obtained, which was not possible using audio excitation with a speaker. Since the beam was focused on the reed, ultrasound stimulated excitation eliminated background effects observed during mechanical shaker excitation, such as vibrations of clamps and supports. We will discuss the results obtained using single, dual, and confocal ultrasound transducers in AM and unmodulated CW modes, along with results obtained using a mechanical shaker and audio excitation using a speaker.

8:30

2aMUa3. Measurement of sound intensity near flue organ pipes. Matthias Scholz, Essa Ibrahim Abdugelder (Chalmers Rm. Acoust. Group, Appl. Acoust., Chalmers, Gothenburg SE-41296, Sweden), Mendel Kleiner (Rensselaer Polytech. Inst., Troy, NY 12180), and Munetaka Yokota (Gothenburg Univ., Gothenburg, Sweden)

Organ voicers sometime experience that organ pipes, within an organ case, acoustically couple to one another. The coupling can be not only between a pipe and its mirror image in the case wall but also between adjacent pipes tuned to the same frequency. This paper reports on intensity measurements done to see the influence of a nearby pipe on the intensity field from a sounding pipe. The sounding pipe was driven pneumatically, and the sound intensity was measured in a plane along the length axis of the pipe. The secondary pipe was positioned, in parallel at different distances, alongside the sounding pipe. The results indicate only a fairly small influence by the secondary pipe on the intensity map of the sounding pipe. [Work supported by the Chalmers Foundation.]

8:45

2aMUa4. The application of cochlear audio analysis techniques to electro-acoustic music. John A. Mills III (Elec. and Computer Eng. Dept., Univ. of Texas, 1 University Station 0803, Austin, TX 78712, nodog@mail.utexas.edu)

Electro-acoustic music has been ignored by music theorists for years. One theory suggests that this deficiency is due to the lack of an objective visual representation of this type of music. This research is focused upon the audio analysis involved in the creation of a "pseudoscoring" for electro-acoustic music. Computer analysis of music is a complicated task. The "holy grail" of automatic computer music analysis has most often been the translation of a continuous pressure variation into traditional Western musical notation. Since electro-acoustic music is rarely able to be transcribed into this type of notation, the automatic analysis of electro-acoustic music confounds many previous algorithms. A top-down approach is suggested in order to extract acoustic and musical information from recordings of electro-acoustics music. A top-down approach has already proven successful in extraction of musical tempo [E. Scheirer, M.I.T. Ph.D. thesis (2000)]. Since humans are able to translate a continuous pressure variation into useful acoustic information, when using a top-down approach to automatic music analysis, a cochlear model is used as a

front end. Although this model apparently encodes some redundant information, cross- and auto-correlation techniques allow easier extraction of some acoustic information. This presentation will detail the current state of this research.

9:00

2aMUa5. Tuning and rotating nearly degenerate modes in handbells. John R. Buschert (Goshen College, 1700 S. Main, Goshen, IN 46526) and Jennifer L. Springer (The Univ. of California, Davis, Davis, CA 95616)

We have studied nearly degenerate pairs of vibrational modes in a C4 handbell. Manufacturers seek to eliminate the beating of these modes by careful choice of the clapper position but this can't always be done satisfactorily. Using electronic holography, we are studying the effects of small, localized mass changes on the frequency and orientation of these mode pairs. One expects one mode of the pair to rotate until it has an anti-node at the point of the added mass and the other to rotate until it has a node there. Then adding further mass should only change the frequency of the first mode. We find some mode pairs respond in just the expected manner but others behave more strangely. [Work supported by an NSF MRI grant.]

9:15

2aMUa6. Low-cost coding of directivity information for the recording of musical instruments. Jonas Braasch, William L. Martens, and Wieslaw Woszczyk (Faculty of Music, McGill Univ., 555 Sherbrooke St. W., Montreal, QC H3A 1B9, Canada, braasch@music.mcgill.ca)

Most musical instruments radiate sound according to characteristic spatial directivity patterns. These patterns are usually not only strongly frequency dependent, but also time-variant functions of various parameters of the instrument, such as pitch and the playing technique applied (e.g., plucking versus bowing of string instruments). To capture the directivity information when recording an instrument, Warusfel and Misdariis (2001) proposed to record an instrument using four channels, one for the monopole and the others for three orthogonal dipole parts. In the new recording setup presented here, it is proposed to store one channel at a high sampling frequency, along with directivity information that is updated only every few milliseconds. Taking the binaural sluggishness of the human auditory system into account in this way provides a low-cost coding scheme for subsequent reproduction of time-variant directivity patterns.

9:30

2aMUa7. Acoustical studies of the American reed organ. James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402)

The reed organ enjoyed a period of great popularity in North America which reached a peak in the late 19th century, when thousands of instruments per year were manufactured and sold in the United States and Canada. Displaced by the emergence of the upright piano, the reed organ had very much fallen out of favor by 1929. In the past decade a number of acoustical investigations have been undertaken on the instrument known as the American reed organ. Observations of reed motion and velocity have been made with electronic proximity sensors and a laser vibrometer system. The variation of the frequency and amplitude of reed vibration as a function of blowing pressure has been explored in some detail and the results compared with predictions of a simple theoretical model. Measurements have been made of the spectrum of the near-field sound including the effects of changes in dimensions of the reed cell. While most treatments of free reed oscillation approximate the reed vibration as a sinusoidal oscillation of a cantilever beam in the fundamental transverse mode, recently some evidence of higher transverse modes and torsional modes of vibration have been observed.

2aMUa8. Calculation of impulse responses with a cellular automata algorithm. Ana Barjau (Dept. of Mech. Eng., Polytech. Univ. of Catalunya, Diagonal 647, 08028 Barcelona, Spain, ana.barjau@upc.es)

The air columns in musical instruments usually have a predominant dimension and thus are very often modeled as 1D systems where uniparametric waves propagate. Different algorithms can be found in the literature to simulate this propagation. The more widely used are finite difference schemes and delay lines. A finite difference scheme (FD) is a numerical integration of a differential formulation (the wave equation), while delay lines (DL) use analytical exact solutions of the wave equation over finite lengths. A new and different approach is that of a cellular automaton (CA) scheme. The underlying philosophy is opposite those of FD and DL, as the starting point is not the wave equation. In a CA approach, the phenomenon to be studied is reduced to a few simple physical laws that are applied to a set of cells representing the physical system (in the present case, the propagation medium). In this paper, a CA will be proposed to obtain the impulse response of different bore geometries. The results will be compared to those obtained with other algorithms.

2aMUa9. Transformations and invariances in equally tempered scales. Alpar Sevgen (Dept. of Phys., Boğaziçi Univ., Bebek 34342, Istanbul, Turkey)

In this work the invariance of the number of accidentals in equally tempered scales (ETSs) under various transformations are proven *analytically*. ETSs with N semitones, M notes and interval structures $\mathbf{n} = \{n_1, n_2, \dots, n_M\}$, where n_i is the number of semitones between adjacent notes, form multiplets of N scales. Member scales of a multiplet are ordered and indexed by scale labels $c_k = kM$ reduced modulo (N) , where $k = 0, 1, \dots, N-1$. Ordered members of the multiplets having the interval structures $\{C^j \mathbf{n}\}$ and $\{C^\ell \mathcal{M} \mathcal{I} \mathbf{n}\}$ (where C is a cyclical permutation, $j, \ell = 0, 1, \dots, M-1$, and $\mathcal{M} \mathcal{I}$ is a mirror image or flip operation on \mathbf{n}) are shown to have the *same* number of sharps and the *same* number of flats, $\mathcal{N}_\sigma(c_k, C^j \mathbf{n}) = \mathcal{N}_\sigma(c_k, C^\ell \mathcal{M} \mathcal{I} \mathbf{n})$ (where $\sigma = \#$ or b , for all j, ℓ and for any scale label c_k). Complementary scales \mathbf{n}_C related to \mathbf{n} by a note \leftrightarrow no-note transformation (e.g., major \leftrightarrow pentatonic), however, have scale labels with opposite signs, and the number of sharps and flats are interchanged $\mathcal{N}_\#(c_k, \mathbf{n}) = \mathcal{N}_b(-c_k, \mathbf{n}_C)$. This explains the degeneracy of the complexity parameter defined as the sum of the number of sharps and flats, proposed as a measure for ETS, explored numerically and reported earlier.

TUESDAY MORNING, 25 MAY 2004

CONFERENCE ROOM K, 10:30 TO 11:55 A.M.

Session 2aMUb

Musical Acoustics: Musical Instruments Developed During the ASA Era

Paul A. Wheeler, Cochair
1595 North 1600 East, Logan, Utah 84341

Ian M. Lindevald, Cochair
Science Division, Truman State University, Kirksville, Missouri 63501

Chair's Introduction—10:30

Invited Papers

10:35

2aMUb1. Musical instrument technology of the 20th century. Paul Wheeler (Utah State Univ., 4120 Old Main Hill, Logan, UT 84322, paul.wheeler@ece.usu.edu)

This paper presents a brief history of the technical development of musical instruments during the 20th century. Starting with early electronic instruments (such as the Theremin—1917) invented prior to the organization of ASA, the history includes the development of electronic organs, synthesizers, and computer music. This paper provides an introduction to the session, giving a framework for the papers which follow in the session.

10:55

2aMUb2. The Electronic Valve Instrument (EVI), an electronic musical wind controller for playing synthesizers. Nyle A. Steiner (269 E. 300 South, Provo, UT 84606, lenyr@earthlink.net)

The Electronic Valve Instrument (EVI) is an electronic musical wind instrument with playing techniques similar to that of a trumpet. Invented by Nyle Steiner in the early 1970's, it was designed to give the performer control of dynamics from breath pressure and the ability to make a humanly generated vibrato. Other musical parameters can be controlled as well. It has a playing range of seven octaves (similar to that of a piano). When musical lines are played using this instrument (controller) connected to an electronic music synthesizer, the sound is much more natural sounding and expressive than when a normal musical keyboard is used. The evolution of this instrument from the pre-Midi era to its latest Midi configuration, principles of operation, synthesizer programming, and its wide use in movie and TV scoring will be discussed. The EVI has played featured musical lines in many major movie soundtracks and TV shows such as *Apocalypse Now*, *Witness*, *Dead Poets Society*, *Fatal Attraction*, *No Way Out*, *Gorillas in the Mist*, and many others. The EVI design has also been adapted as an Electronic Woodwind Instrument (EWI) by Nyle Steiner and has been manufactured and sold worldwide by the AKAI Co. in Japan.

11:15

2aMUb3. New paradigms for musical control—A decade of development at the MIT Media Lab. Joseph A. Paradiso (MIT Media Lab, E15-327, 20 Ames St., Cambridge, MA 02139, joep@media.mit.edu)

As electronic musical instruments liberate the action and energy of control from physical sound production, they are free to mutate into many different forms—the constraints on instrument design have shifted from physics to ergonomics, applications, and aesthetics. Low-cost sensors enable stimuli of all types to act as input, and with a computer interposed between action and sound production, essentially any sonic or musical dynamic can be mapped onto any gesture or activity with an increasingly high degree of interpretation or “mapping.” Accordingly, the notion of a musical instrument is being redefined, and as possibilities broaden, some researchers and artists are striving to break boundaries while others work to quantify and understand expanded metrics for musical interaction. Over the past decade, the author and his colleagues have adapted a wealth of sensor technologies and developed many interaction paradigms to scratch away at the evolving frontier of electronic musical instruments [J. Paradiso, “Electronic music interfaces: new ways to play,” *IEEE Spectrum* **34**(12), 18–30 (1997)]. This presentation will review the status of electronic music controllers, provide a snapshot of current issues that the field is facing, and present various examples of new musical interfaces developed at the MIT Media Lab.

11:35

2aMUb4. Exploring a Van der Pol woodwind. Ana Barjau and Meritxell Genesc (Dept. of Mech. Eng., Polytech. Univ. of Catalunya, Diagonal 647, 08028 Barcelona, Spain, ana.barjau@upc.es)

The behavior of woodwinds can be studied from a theoretical point of view with a variety of mathematical models, ranging from very realistic to fairly simple ones. All of them have in common a nonlinearity responsible for the self-sustained oscillation. In simple models, this nonlinearity is represented through an algebraic equation, while in realistic ones it may be split into different differential equations representing the dynamics of the mechanical device (reed, lips . . .) and of the air flowing through it. In this paper, an intermediate model is explored. A Van der Pol oscillator is used to represent just the nonlinear device of a woodwind (reed valve, lips . . .), while the air column is simplified into a cylindrical or conical one, and its behavior is described through an impulse response. The global system is thus represented mathematically through two coupled differential equations, a linear one containing delayed variables and a nonlinear one. Their integration leads to different sounds, some of which are close to those produced by real instruments.

TUESDAY MORNING, 25 MAY 2004

VERSAILLES TERRACE, 8:35 TO 11:40 A.M.

Session 2aNS

Noise and Psychological and Physiological Acoustics: Noise Impact Evaluation: Old and New I

Brigitte Schulte-Fortkamp, Cochair

Institute of Technical Acoustics, Technical University Berlin, Secr TA 7, Einsteinufer 25, 10587 Berlin, Germany

Klaus Genuit, Cochair

HEAD Acoustics GmbH, Eberstrasse 30a, Herzogenrath 52134, Germany

Chair's Introduction—8:35

Invited Papers

8:40

2aNS1. Relation between loudness and annoyance over time: Implications for assessing the perception of low-frequency noise.

Rhona Hellman (Inst. of Hearing, Speech, & Lang. and SLPA Dept. [106A FR], Northeastern Univ., 360 Huntington Ave., Boston, MA 02115-5000, hellman@neu.edu) and Norm Broner (Vipac Engineers & Scientists Ltd., Melbourne, Australia)

The literature suggests that, in contrast to loudness, annoyance increases over time. However, in these early measurements the annoyance exposure time usually did not exceed 120 s. In the current work, this temporal interval was extended to 3600 s (1 h). Within this time frame, both loudness and annoyance of low-frequency noise spectra containing dominant tones were measured at 12 temporal intervals by the method of successive magnitude estimation. Four noise spectra that evoked the largest annoyance-to-loudness ratios in previous reports were selected as stimuli. On average, after 1 h loudness declined by 8% and annoyance declined by 12.5% at moderate sensation levels (SL). By comparison, at 27-dB SL annoyance initially exceeded loudness by 51% up to 120 s and then, declined with time more slowly than loudness. These results indicate that, despite an 11-phon decrease in loudness level, the annoyance of a low-frequency steady noise with dominant tones below 50 Hz may initially increase over time. The annoyance increase may then be followed by a gradual annoyance decrease that mirrors the loudness decrease but at a slower rate. Taken together, the obtained relations imply, in accord with other data, that annoyance is not solely loudness based.

9:00

2aNS2. Relationships of loudness, level, and time structure in pipe organ registrations and design. Wade Bray (HEAD acoustics, Inc., 6964 Kensington Rd., Brighton, MI 48116)

Time-dependent loudnesses and sound pressure levels were measured binaurally from a variety of individual stops and multi-stop registrations of several classical and theater pipe organs covering a wide range of sizes, tonal designs, pipe scalings, wind pressures, and spatial relationships. Studies were also made of modulation in individual low-frequency notes, and in notes and chords affected by the presence or absence of tremulants. Timbres of certain solo stops were assessed in terms of the relationship of specific loudness and sound pressure spectra to their importance in creating subjective timbre. In addition to loudness and sound pressure measurement, advanced analysis by a hearing model (Sottek) and a temporal/tonal pattern-measurement algorithm (“relative approach”) were employed. Significant differences in the relationship of loudness and level were found depending principally on tonal design, pipe scaling and wind pressures on one hand, and octave-related tonal centers and timbral characteristics of individual stops on the other hand. Consideration was also given to the subjective spatial soundscape resulting from different pipe organ physical layouts and from simultaneous registrations between or among physically-separated speaking divisions.

9:20

2aNS3. Spatial representation of soundscape. Mohammed Boubezari and Jos-Luis Bento Coelho (CAPS, Instituto Superior Tecnico, 1049-001 Lisboa, Portugal, m.boubezari@ist.utl.pt)

For the last 30 years the concept of soundscape has been largely adopted in many scientific disciplines and by the urban experts for the benefit of a better comprehension and management of the sound environment. However, the spatial representation of the soundscape as a simple tool for the description, management or composition of sound environment is always needed. In this article a method is presented for the spatial sound representation with differentiated sources. The first results are shown. This method gives an account of the soundscape as close as possible to the way it can be perceived by the listener in each location. This method generates qualitative sound maps in a reduced urban scale, based on *in situ* measurements and on the implication of the measuring subject perception. The maps are sufficient enough to isolate many sound sources of the overall sound field. In this manner, sound quality refers to the sound attribute of a perceived object. It is neither an aesthetic judgment nor traditional psychoacoustics criteria. Concrete examples of application to squares in the city of Lisbon will be shown and discussed. The limits and the prospects of such a qualitative representation will also be presented and discussed.

9:40

2aNS4. Listening to a town: The urban soundscape as an image of the city. Catherine Semidor (GRECO-Bx, EAPBx, Domaine de Raba, F-33400 Talence, France, catherine.semidor@bordeaux.archi.fr)

City dwellers often have a visual representation of their town. They are able to identify some areas from pictures. But, urban sounds are also a characteristic part of the city's identity. A previous paper [C. Semidor, Proceedings 20th PLEA, Santiago, Chile (2003)] shows the possibility to distinguish the soundscape of a “car-free” Sunday from the soundscape of a Sunday with cars. The proposed method based on the so-called soundwalk could be very illustrative for this suggestion. These soundwalks are sound recordings made with a “dummy” head and a DAT recorder, on walks following a trajectory with different urban forms and, in this case study, in different towns where a tramway is running. The acoustical images (time versus frequency graphs) of the sound signals could be a representation of these soundscapes. This type of information could help urban planners and other town designers to improve the acoustic comfort of the cities. The objective of this approach is to valorize what is pleasant and agreeable in the urban sound environment.

10:00–10:20 Break

10:20

2aNS5. Context sensitive road traffic noise impact mapping—Taking the neighborhood soundscape into account. Ronny Klboe (Inst. of Transport Economics, Grensesvingen 7, N-0602 Oslo, Norway), Erik Engelién, and Margrete Steinnes (Statistics Norway, N-2225 Kongsvinger, Norway)

Road traffic noise exposure contour maps are difficult to interpret by nonexperts who are familiar with neither the road traffic noise exposure measures nor their associated impacts. An alternative is to map impacts such as annoyance. However, in urban areas the noise impacts are multi-factorially determined and context sensitive. In particular people become more annoyed by a given noise level at the most exposed facade of their dwelling when their neighborhood soundscape is even noisier. A two-tiered approach makes use of contextual soundscape information in determining noise impacts and builds contiguous neighborhood sonoscapes delimiting neighborhood areas with similar noise impacts. Neighborhood sonoscape maps facilitate a more precise targeting of local noise abatement measures, and can illustrate the impacts of noise abatement measures. With appropriate classification and class labels neighborhood sonoscapes provide an environmental labeling of the expected perceived sound quality of the neighborhood for consumers, the public and planners. Neighborhood sonoscape maps may be utilized for national stratification and subsequent two-stage cluster sampling of the population. The advantage of this approach is that focused traffic counts, extended sound modeling and monitoring of noise abatement procedures, population composition, etc. can be undertaken for a limited representative set of neighborhood sonoscapes.

10:40

2aNS6. How to boost soundscapes with tax breaks—On the application of Henry George’s environmental economics to noise annoyance. Cay Hehner (Inst. of Tech. Acoust., TU-Berlin, Einsteinufer 25, D-10587 Berlin, Germany, kskanda@mach.ut.tu-berlin.de)

The improvement of the characteristic soundscape in a given metropolitan area depends largely on a varied number of parameters not the least of which are economic. The way to sustain a large city not only economically responsible behavior patterns and activities are required, but ecologically viable ones as well. And, not coincidentally, if both patterns come into balance, a reduction of noise annoyance ensues. The cohesive set of measures that would provide the framework to ensure that the necessary economic activity is not antagonistic to (acoustic) ecology is provided by the US economist Henry George who advocated a comprehensive tax on land and all natural resources to replace eventually the bulk of all taxes checking production. New data will be provided in this paper to show the results of the application of a Georgist approach of eco-taxation to the improvement of sound quality and the reduction of noise annoyance.

11:00

2aNS7. Assessing noise pollution in sensitive areas: Soundscape analysis in an alpine valley by psychoacoustic means. Peter Lercher (Univ. of Innsbruck, Sonnenburgstrasse 16, A-6020 Innsbruck, Austria, Peter.Lercher@uibk.ac.at), Klaus Genuit (HEAD acoustics GmbH, D-52120 Herzogenrath, Germany), Urs Reichart (Inst. of Tech. Acoust., D-10587 Berlin, Germany), and Dietrich Heimann (Institut fuer Physik der Atmosphaere, D-82234 Wessling, Germany)

Alpine valleys are sensitive areas due to topography, meteorology, housing, and land-use pattern, that modify noise propagation and make protection against noise pollution rather difficult. The “amphitheater” effect was mentioned as explanation for deviating noise-annoyance curves and health effects observed at lower sound levels. However, detailed empirical analyses are lacking. In this study a series of simultaneous, binaural sound recordings was carried out in several cross sections of the Wipp Valley along the central-European transportation route to the Brenner pass. During 6 weeks a wide variety of day- and week times was sampled with variable wind (“Foehn”) and weather conditions (dry, rain, snow). Sound recordings were paralleled by meteorological recordings near the source and on the slope. First analyses have revealed several facts. (1) The assumption of linear sound propagation to the slope is seriously in error. (2) Tonal components from gearboxes are a significant feature in the slope recordings. (3) Low-frequency modulations make the sound more intrusive on the slope—while near the source this feature is better masked. (4) Low background sound levels (<30 dB,A) on the slopes are in sharp contrast with incoming sound levels (52 dB,A, L_{eq} about 1200 m from the source). (5) Meteorology leads to substantial changes in measured sound levels.

11:20

2aNS8. Transfer of knowledge from sound quality measurement to noise impact evaluation. Klaus Genuit (HEAD acoustics, Ebertstr. 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

It is well known that the measurement and analysis of sound quality requires a complex procedure with consideration of the physical, psychoacoustical and psychological aspects of sound. Sound quality cannot be described only by a simple value based on A-weighted sound pressure level measurements. The A-weighted sound pressure level is sufficient to predict the probability that the human ear could be damaged by sound but the A-weighted level is not the correct descriptor for the annoyance of a complex sound situation given by several different sound events at different and especially moving positions (soundscape). On the one side, the consideration of the spectral distribution and the temporal pattern (psychoacoustics) is requested and, on the other side, the subjective attitude with respect to the sound situation, the expectation and experience of the people (psychology) have to be included in context with the complete noise impact evaluation. This paper describes applications of the newest methods of sound quality measurements—as it is well introduced at the car manufacturers—based on artificial head recordings and signal processing comparable to the human hearing used in noisy environments like community/traffic noise.

Session 2aPA

Physical Acoustics: Materials Characterization

Ben Dzikowicz, Chair

Coastal Systems Station, 6702 West Highway 98, Panama City, Florida 32407-7001

Contributed Papers

9:30

2aPA1. Nondestructive evaluation of multilayered structures using ultrasonic guided waves generated by an electromagnetic acoustic array transducer. Michael Quarry (Lawrence Livermore Natl. Lab., 7000 East Ave., Livermore, CA 94550, quarry1@llnl.gov)

Multilayered structures with adhesively bonded plates are a common inspection problem for nondestructive evaluation. In many cases, large areas must be inspected with limited access. One powerful method for inspecting these structures is to excite ultrasonic guided wave modes that propagate over distances of many meters and into inaccessible areas that may be embedded within another structure. The utilization of guided wave modes with horizontal shear displacement for the nondestructive evaluation of adhesively bonded plates is studied here. Modes were excited in the frequency region of 200 to 800 kHz by an electromagnetic acoustic array transducer with half-inch and quarter-inch spacing. Samples were constructed for testing that included multilayered plates as well as multilayered structures with a transition between a single layer plate and a multilayer plate. Notches of various sizes were placed in the bottom layer to simulate cracks and test sensitivity. Experimental data shows that an electromagnetic acoustic array was able to generate a dominant horizontal shear mode to detect the notches with sufficient signal to noise for practical inspection by tuning the frequency of excitation. Notches of 10% through-wall were detected.

9:45

2aPA2. Linear and nonlinear acoustic control of accumulated fatigue damage in steel. Alexander Sutin (Stevens Inst. of Technol., Hoboken, NJ), Yulian Kin (Purdue Univ. Calumet, Hammond, IN, asutin@stevens.edu), and Paul Johnson (Los Alamos Natl. Lab., Los Alamos, NM)

We present results of linear and nonlinear acoustic testing of steel samples with different levels of fatigue damage. The steel specimens were tested under programmed cyclic loading on a fatigue testing machine and accumulated different levels of fatigue damage. No visible surface-crack formations during fatigue cycling were observed. In other words, the emphasis was placed on the characterization of continued but physically invisible in service life conditions damage in different materials and structures. Both linear and nonlinear acoustic techniques were used to assess damage accumulation. (1) Impulse resonant acoustic spectroscopy (IRAS) is based on analysis of the free-sample vibration after impact excitation. It demonstrated the increasing of the resonance frequencies and Q factor with damage accumulation. (2) Nonlinear resonant acoustic spectroscopy (NRAS) is based on measurement of the resonance response for different levels of acoustic excitation. The amplitude-dependent frequency shift for damaged steel was observed to increase with damage accumulation. (3) Nonlinear wave modulation spectroscopy (NWMS) implies the modulation of ultrasound wave by lower frequency vibration. High level of the side-band components for damaged samples were observed. The comparison of different methods is given.

10:00

2aPA3. Ultrasonic attenuation in Voronoi polycrystals. Joseph A. Turner and Goutam Ghoshal (Dept. of Eng. Mech., Univ. of Nebraska-Lincoln, W317.4 Nebraska Hall, Lincoln, NE 68588-0526, jturner@unl.edu)

Studies of elastic wave scattering in polycrystalline media are applicable to ultrasonic materials characterization and nondestructive evaluation. Ultrasonic attenuation is used to extract microstructural parameters such as grain size and grain texture. The relations between experimental data and microstructural information require ultrasonic scattering models that often oversimplify real microstructures. Here, these relations are examined in terms of numerical simulations of elastic waves through two-dimensional polycrystals. The numerical models, based on the Voronoi polycrystal, have been shown to model accurately the microstructure of polycrystalline metals and ceramics. Methods of creating Voronoi polycrystals with appropriate microstructure, including grain elongation and grain orientation, are first discussed followed by discussions of the wave propagation. The wave simulations are based on the discretization of these Voronoi cells, using finite elements, which are then integrated directly in time using ABAQUS/EXPLICIT software. The material properties of the individual Voronoi cells are chosen according to appropriate material distributions. Here, the focus is on cubic crystals that are either statistically isotropic or transversely isotropic. Attenuation results for longitudinal waves are presented and compared with scattering theories for materials of common interest. These simulations provide insight into the attenuation models relevant for ultrasonic measurements involving polycrystalline materials.

10:15–10:30 Break

10:30

2aPA4. Using eccentricity in multilayered scatterer as mechanism for tunable phononic materials. Liang-Wu Cai (Dept. of Mech. and Nuclear Eng., Kansas State Univ., 331 Rathbone Hall, Manhattan, KS 66506, cai@ksu.edu)

The concept of multiple scattering in single scatterers, originally proposed by the author, can be utilized to analyze the scattering of elastic waves by scatterers having multiple eccentric layers. Using a dual-layer circular eccentric cylindrical scatterer as an example, it is demonstrated that, in a certain frequency range, the geometric eccentricity can significantly influence the pattern of the scattered field. The geometric eccentricity of such a scatterer can be described by three parameters: the orientation angle between the line connecting the two centers and the incident direction; the ratio of radii of the two layers, and the ratio of the distance between two centers and the radius of the outer layer. It is found that, when the wavelength is comparable to the outer diameter of the scatterer, the pattern of the scattered wave field is extremely sensitive to all three geometric parameters. Furthermore, this is the frequency range that is very close to the band gap as dictated by the lattice structure. It is expected that

when such multilayered scatterers are arranged in a lattice pattern, the eccentricity can be exploited for tuning the band-gap characteristics of the resulting phononic material.

10:45

2aPA5. Active reduction of acceleration sensitivity of quartz thickness shear resonators. Mihir Patel and Yook-Kong Yong (Rutgers Univ., 623 Bowser Rd., Piscataway, NJ 08854)

The piezoelectric quartz thickness shear resonator is a very important component in stable oscillators. The resonator is usually a rectangular or circular plate with electrodes on the top and bottom surfaces. When the resonator is subjected to acceleration, the inertial effects cause stresses and strains in the plate which in turn cause the resonant frequency to shift. A method is proposed to actively minimize the inertial stresses and strains by piezoelectric stiffening using a dc electric field bias across the electrodes. Finite element models are created to calculate the effects of acceleration on resonant frequency and compare the results with experimental data. The models are then used to calculate the effects of the dc electric field bias and demonstrate the reduction of acceleration sensitivity. Results are shown for the AT- and SC-cut quartz resonators.

11:00

2aPA6. Magneto-elastic interactions in Fe–Ga alloys. G. Petculescu, H. E. Bass (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, 1 Coliseum Dr., Univ., MS 38677), T. A. Lorasso (Ames Lab., Ames, IA 50011-3020), and A. E. Clark (Clark Assoc., Adelphi, MD 20783)

The elastic anisotropy of iron's most common α -structure increases significantly when gallium is substituted for a fraction of the iron. Concurrently occurring, the tremendous boost in tetragonal magnetostriction [one order of magnitude above that of pure Fe, $\frac{3}{2}\lambda_{100}(\text{Fe}_{81}\text{Ga}_{19})$

=400 ppm] makes Fe–Ga alloys important for magnetoelastic applications. Measurements of the elastic tensor of Fe–Ga single crystals, with or without a magnetic field and within 4–300 K, were performed using the resonant ultrasound spectroscopy technique on 100-cut rectangular parallelepipeds. In the vicinity of 24% Ga an anomaly occurs in the magnetostriction as well as in the elastic moduli and in the magnitude of their ΔE -effect, in an apparent conspiracy to maintain the magnetoelastic energy-constants ($b_{1,2}$) "normal." [Work supported by ONR.]

11:15

2aPA7. Acousto-domain interaction in ferroelectric lithium niobate. Igor V. Ostrovskii (Dept. of Phys. and Astron., Univ. of Mississippi, University, MS 38677, iostrov@phy.olemiss.edu), Mukola M. Borovoy, Oleg O. Korotchenkov, Andriy B. Nadochii, and Roman G. Chupryna (Kiev Shevchenko Univ., Kiev 02022, Ukraine)

First observation of the reorientation of ferroelectric domains in the lithium niobate single crystals induced by ultrasound is reported. This effect is detected from the samples, which are treated by megahertz frequency-range ultrasound with above certain threshold amplitude at room temperature. Acoustic strain amplitude causing the domain reorientation is of the order of $E10^{-5}$. The effect is directly revealed by chemically etched crystal surfaces, and independently confirmed by the acoustically induced evolution of x-ray diffraction rocking curves and changes in acousto-electric properties of the lithium niobate crystals. The physical mechanism responsible for the interaction of the domains and ultrasound can be attributed to the mechanical stress and piezoelectric field produced by a piezo-active acoustic wave. A new effect of acousto-domain interaction in ferroelectric crystals might be taken into account for a wide variety of fundamental physical phenomena involving propagation of the acoustic waves in real crystals.

TUESDAY MORNING, 25 MAY 2004

ROYAL BALLROOM B, 8:00 A.M. TO 12:00 NOON

Session 2aPP

Psychological and Physiological Acoustics: Perceptual Organization of Sound (Lecture/Poster Session)

Jennifer Lentz, Chair

Speech and Hearing Sciences, Indiana University, 200 South Jordan Avenue, Bloomington, Indiana 47405

Chair's Introduction—8:00

Invited Papers

8:05

2aPP1. A really complicated problem: Auditory scene analysis. William A. Yost (Parmly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626)

It has been more than a decade since Al Bregman and other authors brought the challenge of auditory scene analysis back to the attention of auditory science. While a lot of research has been done on and around this topic, an accepted theory of auditory scene analysis has not evolved. Auditory science has little, if any, information about how the nervous system solves this problem, and there have not been any major successes in developing computational methods that solve the problem for most real-world auditory scenes. I will argue that the major reason that more has not been accomplished is that auditory scene analysis is a really hard problem. If one starts with a single sound source and tries to understand how the auditory system determines this single source, the problem is already very complicated without adding other sources that occur at the same time as is the typical depiction of the auditory scene. In this paper I will illustrate some of the challenges that exist for determining the auditory scene that have not received a lot of attention, as well as some of the more discussed aspects of the challenge. [Work supported by NIDCD.]

8:25

2aPP2. Computer models of possible physiological contributions to low-level auditory scene analysis. Ray Meddis (Dept. of Psych., Essex Univ., Colchester, Essex CO3 4SQ, UK)

Auditory selective attention is a general term for a wide range of phenomena including grouping and streaming of sound sources. While we know a great deal about the circumstances in which these phenomena occur, we have little understanding of the physiological mechanisms that give rise to these effects. Because attention is sometimes under conscious control, it is tempting to conclude that attention is a high-level/cortical function and beyond our current understanding of brain physiology. However, a number of mechanisms operating at the level of the brainstem may well make an important contribution to auditory scene analysis. Because we know much more about the response of the brainstem to auditory stimulation, we can begin some speculative modeling concerning their possible involvement. Two mechanisms will be discussed in terms of their possible relevance: lateral and recurrent inhibition at the level of the cochlear nucleus. These are likely to contribute to the selection of auditory channels. A new approach to within-channel selection on the basis of pitch will also be discussed. These approaches will be illustrated using computer models of the underlying physiology and their response to stimuli used in psychophysical experiments.

8:45

2aPP3. The perceptual organization of complex sounds by birds. Micheal L. Dent (Dept. of Physiol., 1300 University Ave., Univ. of Wisconsin, Madison, WI 53706, dent@physiology.wisc.edu)

Birds have proven to be ideal models for the perceptual organization of complex sounds because, like humans, they produce, learn, and use complex acoustic signals for communication. Although conducted in laboratory settings, measures of auditory abilities in birds are usually designed to parallel the acoustic problems faced in their natural habitats, including the location of conspecifics, discrimination among potential mates, prey localization, predator avoidance, and territorial defense. As a result, there is probably more known about hearing in birds under both natural and laboratory conditions than in any other nonhuman organism. Behavioral and/or physiological experiments on complex sound perception in birds have revealed that they exhibit serial pattern perception, can discriminate frequency changes in tones embedded within tonal patterns regardless of stimulus uncertainty conditions, segregate signals into auditory streams, and exhibit comodulation masking release. In addition, binaural experiments have revealed that birds exhibit both the cocktail party effect and the precedence effect. Taken together, these results suggest that, like humans, auditory scene analysis plays a general role in auditory perception in birds and probably other animals that must parse the world into auditory objects. [Work supported by NIH DC006124.]

9:05

2aPP4. Sensorineural hearing loss and auditory perceptual organization. Joseph W. Hall III, John H. Grose, and Emily Buss (Univ. of North Carolina at Chapel Hill, 130 Mason Farm Rd., CB 7070, 1115 Bioinformatics Bldg., Chapel Hill, NC 27599-7070, jwh@med.unc.edu)

This talk will consider the implications of sensorineural hearing loss for auditory perceptual organization. In everyday environments, the listener is often faced with the difficulty of processing a target sound that intermingles acoustically with one or more extraneous sounds. Under such circumstances, several auditory processes enable the complex waveforms reaching the two ears to be interpreted in terms of putative auditory objects giving rise to the target and extraneous sounds. Such processes of perceptual organization depend upon the central analysis of cues that allow distributed spectral information to be either linked together or split apart on the basis of details related to such variables as synchrony of onset/modulation, harmonic relation, rhythm, and interaural differences. Efficient perceptual organization must depend not only upon such central auditory analyses but also upon the fidelity with which the peripheral auditory system encodes the spectral and temporal characteristics of sound. We will consider the implications of sensorineural hearing loss for perceptual organization in terms of both peripheral and central auditory processes.

9:25

2aPP5. Auditory grouping in the perception of speech and complex sounds. Chris Darwin and Marie Rivenez (Dept of Psych., Univ. of Sussex, Brighton BN1 9QG, UK, cjd@biols.susx.ac.uk)

This talk will give an overview of experimental work on auditory grouping in speech perception including the use of grouping cues in the extraction of source-specific auditory information, and the tracking of sound sources across time. Work on the perception of unattended speech sounds will be briefly reviewed and some recent experiments described demonstrating the importance of pitch differences in allowing lexical processing of speech on the unattended ear. The relationship between auditory grouping and auditory continuity will also be discussed together with recent experiments on the role of grouping in the perceptual continuity of complex sounds.

Contributed Papers

9:45

2aPP6. Binaural release from informational masking: Results from a speech identification task. Frederick J. Gallun, Christine R. Mason, and Gerald Kidd, Jr. (Hearing Res. Ctr. and Commun. Disord., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215)

This study examined how binaural cues can reduce informational masking (IM) in a speech identification task. Target and masker sentences were processed into non-overlapping frequency bands, thus limiting "energetic masking," and were presented over headphones. Listeners identi-

fied key words in the target sentence. In a baseline condition, target and masker were presented monotonically (SmMm), producing large amounts of IM. Binaural release from IM (i.e., improved performance re. SmMm) was observed when the target was presented monotonically and the masker diotically (SmM0), suggesting that the shift of masker image away from target led to the reduction in IM. Creating large interaural differences in level (ILDs) showed that for a monaural target, binaural release, though related to ILD, occurred even when the masker image should have been lateralized at the target ear. For differences in time (ITDs) up to 600 μ sec, the amount of binaural release was completely independent of ITD. For a

diotic target and binaural masker, however, release only occurred for large ITDs or ILDs. These results suggest that for monaural targets and binaural maskers, IM in a speech task can be reduced through a binaural cue that is present across the entire range of biologically plausible ITDs and ILDs.

10:00

2aPP7. Using spatialized sound cues in an auditorily rich environment. Derek Brock, James A. Ballas, Janet L. Stroup, and Brian McClimens (Naval Res. Lab., Code 5513, 4555 Overlook Ave. S.W., Washington, DC 20375)

Previous Navy research has demonstrated that spatialized sound cues in an otherwise quiet setting are useful for directing attention and improving performance by 16.8% or more in the decision component of a complex dual-task. To examine whether the benefits of this technique are undermined in the presence of additional, unrelated sounds, a background recording of operations in a Navy command center and a voice communications response task [Bolia *et al.*, *J. Acoust. Soc. Am.* **107**, 1065–1066 (2000)] were used to simulate the conditions of an auditorily rich military environment. Without the benefit of spatialized sound cues, performance in the presence of this extraneous auditory information, as measured by decision response times, was an average of 13.6% worse than baseline performance in an earlier study. Performance improved when the cues were present by an average of 18.3%, but this improvement remained below the improvement observed in the baseline study by an average of 11.5%. It is concluded that while the two types of extraneous sound infor-

mation used in this study degrade performance in the decision task, there is no interaction with the relative performance benefit provided by the use of spatialized auditory cues. [Work supported by ONR.]

10:15

2aPP8. The probabilistic relationship between auditory stimuli and their natural sources predicts pitch. David A Schwartz and Dale Purves (Ctr. for Cognit. Neurosci., Duke Univ., Box 90999, Durham, NC 27708-0999)

The phenomenology of pitch has been difficult to rationalize and remains the subject of much debate. We here examine the hypothesis that audition generates pitch percepts by relating inherently ambiguous sound stimuli to their probable sources in the human auditory environment. A database of speech sounds, the principal source of periodic sound energy for human listeners, was compiled and the dominant periodicity of each speech sound determined. A set of synthetic test stimuli were used to assess whether the major pitch phenomena described in the literature could be explained by the probabilistic relationship between the stimuli and their probable sources (i.e., speech sounds). The phenomena tested included the perception of the missing fundamental, the pitch shift of the residue, spectral dominance and the perception of pitch strength. In each case, the conditional probability distribution of speech sound periodicities accurately predicted the pitches normally heard in response to the test stimuli. We conclude that pitch depends on an auditory process that relates inevitably ambiguous sound stimuli to their probable natural sources.

Poster Papers

All posters will be on display and all contributors will be at their posters from 10:30 a.m. to 12:00 noon.

2aPP9. Effects of frequency transposition on discrimination of recycled tonal sequences. Tomoko Hashida (Grad. School of Interdisciplinary Information Studies, Univ. of Tokyo, Japan, hashida@hc.t.u-tokyo.ac.jp), Takuro Kayahara and Takao Sato (Univ. of Tokyo, Japan)

Recycled sequences have been used for examining perception of sequences which requires temporal-order judgments of components to eliminate advantages of the initial and the terminal components in sequence recognition. The purpose of this study was to examine whether listeners could discriminate recycled sequences when the center frequency was varied. Each stimulus in this study was a random sequence of ten 200-ms pure tones divided equally in log scale within 2/3 octave. The center frequency of the standard sequence was fixed at 500 Hz, while those of test sequences were varied from 62.5 to 4000 Hz. Two test sequences were prepared for each center frequency. One had the same order as the standard; the other had a different order by reversing two contiguous components. Subjects were asked to judge which one of these two test sequences had the same order as that of the standard sequence. Results indicated that (1) listeners could distinguish the order of sequences even when the center frequency was transposed, and that (2) the range of transposition within which listeners could distinguish was limited within several octaves.

2aPP10. Asymmetrical perception of dynamic changes in rising and falling sounds. Darren Kwong (Dept. of Bioengineering, Univ. of Pennsylvania, Philadelphia, PA 19104) and Fan-Gang Zeng (Univ. of California, Irvine, CA 92697)

Previous studies have demonstrated a perceptual enhancement effect for rising sounds and interpreted at an ecological level the enhancement of rising sounds as a perceptual bias towards an approaching auditory object [Neuhoff, *Nature (London)* **395**, 123 (1998)]. This study has not been independently replicated and both its interpretation and mechanisms are still debatable. The present study was aimed at replicating and extending the Neuhoff study by measuring just-noticeable differences in intensity for

rising tone and falling sounds. Pure tones, harmonic complex tones, and white noise were used as the stimuli. Similar to Neuhoff's study, subjects had to estimate by rating on a scale the loudness changes heard corresponding to rising or falling sounds. In addition, they had to discriminate between either a rising or falling sound and a steady-state sound or discriminate between two rising or falling sounds. Both magnitude estimation and objective discrimination data were consistent with Neuhoff's results, showing that rising sounds needed 2 and 7 dB less than falling sounds in discriminating them from a steady-state sound and from a similarly dynamically changing sound, respectively. Mechanisms related to this perceptual asymmetry will be discussed.

2aPP11. A theory of three-dimensional auditory perception. Kazi Saifuddin (Dept. of Psych., Eden College, Dhaka, Bangladesh, kazi_64@yahoo.com)

A theory of auditory dimensions regarding temporal perception is proposed on the basis of the results found from a series of experiments conducted. In all experiments, relationships were investigated between the subjective judgments and the factors extracted from the autocorrelation function (ACF) of the auditory stimuli. The factors were changed by using different properties of pure-tone, complex-tone, white-noise and bandpass-noise stimuli. Experiments by paired-comparison method were conducted in the sound proof chamber except for one in a concert hall. Human subjects were asked to compare the durations of the two successive stimuli in the pair. Subjective durations were obtained in the psychometric function. Auditory stimuli were selected to use on the basis of the measured factors of ACF as parameters. Obtained results showed significant correlation between the factors of ACF and the subjective durations described well by the theory. The theory indicates loudness and pitch as two fundamental dimensions and whether the third one is the duration of the stimuli.

2aPP12. Auditory stream segregation with multi-tonal complexes in hearing-impaired listeners. Deanna S. Rogers and Jennifer J. Lentz (Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47401, deanroge@indiana.edu)

The ability to segregate sounds into different streams was investigated in normally hearing and hearing-impaired listeners. Fusion and fission boundaries were measured using 6-tone complexes with tones equally spaced in log frequency. An ABA-ABA- sequence was used in which A represents a multitone complex ranging from either 250–1000 Hz (low-frequency region) or 1000–4000 Hz (high-frequency region). B also represents a multitone complex with same log spacing as A. Multitonal complexes were 100 ms in duration with 20-ms ramps, and- represents a silent interval of 100 ms. To measure the fusion boundary, the first tone of the B stimulus was either 375 Hz (low) or 1500 Hz (high) and shifted downward in frequency with each progressive ABA triplet until the listener pressed a button indicating that a “galloping” rhythm was heard. When measuring the fusion boundary, the first tone of the B stimulus was 252 or 1030 Hz and shifted upward with each triplet. Listeners then pressed a button when the “galloping rhythm ended.” Data suggest that hearing-impaired subjects have different fission and fusion boundaries than normal-hearing listeners. These data will be discussed in terms of both peripheral and central factors.

2aPP13. Across-channel interference and grouping. Emily Buss, Joseph W. Hall III, and John H. Grose (Univ. of North Carolina at Chapel Hill, 130 Mason Farm Rd., CB7070, 1115 Bioinformatics Bldg., Chapel Hill, NC 27599, ebuss@med.unc.edu)

Thresholds were measured for detection of an increment in level of a 500-ms, 1-kHz target tone, either in quiet or in the presence of masker tones at 0.5 and 2 kHz. Maskers were either 500 ms in duration, gated synchronously with the target, or 1 s in duration, leading the target by 500 ms. In some conditions the target and/or the maskers were amplitude modulated by multiplication with a raised cosine. The level of the target to which the increment was added was fixed at 60 dB SPL; the level of each masker was selected independently from a uniform distribution ± 8 dB *re*: 60 dB. Thresholds were higher in the presence than in quiet, despite the 1-octave spectral separation. Maximum threshold elevation occurred for synchronous target/masker gating, and with all tones (target and masker) either steady or 10-Hz amplitude modulated. Thresholds in these conditions were on the order of 10 dB [in terms of $10 \log(\Delta I/I)$]. Performance tended to improve with introduction of gating asynchrony and with 10-Hz modulation on either target or maskers. Improvements were on average approximately 5 dB, with more pronounced effects of gating. Results will be discussed in terms of spectral cues and perceptual organization.

2aPP14. Differentiating speech and nonspeech sounds via amplitude envelope cues. Robert J. Lehnhoff, Jr., Winifred Strange, and Glenis Long (Speech and Hearing Sci., City Univ. of New York—Grad. Ctr., 365 Fifth Ave., New York, NY 10016)

Recent evidence from neuroscience and behavioral speech science suggests that the temporal modulation pattern of the speech signal plays a distinctive role in speech perception. As a first step in exploring the nature of the perceptually relevant information in the temporal pattern of speech, this experiment examined whether speech versus nonspeech environmental sounds could be differentiated on the basis of their amplitude envelopes. Conversational speech was recorded from native speakers of six different languages (French, German, Hebrew, Hindi, Japanese, and Russian) along with samples of their English. Nonspeech sounds included animal vocalizations, water sounds, and other environmental sounds (e.g., thunder). The stimulus set included 30 2-s speech segments and 30 2-s nonspeech events. Frequency information was removed from all stimuli using a technique described by Dorman *et al.* [J. Acoust. Soc. Am. **102** (1997)]. Nine normal-hearing adult listeners participated in the experiment. Subjects decided whether each sound was (originally) speech or

nonspeech and rated their confidence (7-point Likert scale). Overall, subjects differentiated speech from nonspeech very accurately (84% correct). Only 12 stimuli were not correctly categorized at greater than chance levels. Acoustical analysis is underway to determine what parameters of the amplitude envelope differentiate speech from nonspeech sounds.

2aPP15. The role of spectral asynchrony in the identification of spectrally smeared environmental sounds. Valeriy Shafiro (Dept. of Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1015 Armour Academic Ctr., 1653 W. Congress Pkwy., Chicago, IL 60612, valeriy_shafiro@rush.edu) and Yana Gilichinskaya (CUNY Grad. Ctr., New York, NY 10016)

This study examined possible contributions of spectral asynchrony caused by the differences in filter group delays across channels to the previously reported decline in identification accuracy of several spectrally smeared environmental sounds [V. Shafiro, J. J. Jenkins, and W. Strange, J. Acoust. Soc. Am. **113**(4), 2326–2327 (2003)]. As in the previous experiment, stimuli were obtained by processing environmental sounds through a noise-based vocoder with a varying number of frequency channels. In the present experiment, however, the group delays were held constant across channels, while the overall frequency response of the filters closely resembled the earlier one. Listeners were instructed to identify the most likely source of each sound by selecting it from 60 possible response options. Results indicated that identification accuracy improved for some environmental sounds, while it worsened for other sounds. Overall, there was little improvement in identification accuracy across channels. These findings are interpreted as reflecting the limitations in signal-processing parameters and undesirable processing artifacts. Preliminary results of a different signal-processing method that maintains a constant group delay and uses linear frequency intervals for individual channels indicate a greater improvement in identification accuracy.

2aPP16. Auditory distance perception in fixed and varying simulated acoustic environments. Matthew Schoolmaster (Hearing Res. Ctr. and Dept. of Cognit. and Neural Systems, Boston Univ., 677 Beacon St., Boston, MA 02215, msch@bu.edu), Norbert Kopčo (Boston Univ., kopco@bu.edu), and Barbara G. Shinn-Cunningham (Boston Univ., Boston, MA 02215)

Listeners must calibrate to the acoustic environment in order to judge source distance using reverberation. Results of a previous study [Schoolmaster, Kopčo, and Shinn-Cunningham, J. Acoust. Soc. Am. **113**, 2285 (2003)] showed that distance perception is more accurate when the simulated acoustic environment is consistent rather than randomly chosen from trial to trial; however, the environments were very different (anechoic versus a large classroom). The study was replicated here using two rooms that were much more similar. Each subject completed two series of trials. In the fixed-room series, the room (large or small) was fixed within a trial block. In the mixed-room series, the room was randomly chosen from trial to trial. Half of the subjects performed the mixed and half performed the fixed series first. Differences between subject groups were smaller in the current study than when anechoic and reverberant trials were intermingled; performance in the two subject groups was similar for (1) fixed trials and (2) trials simulating the large room. However, there was an interaction between subject group and room. Results suggest that listeners calibrate distance judgments based on past experience, but that such calibration partially generalizes to similar environments [Work supported by AFOSR and the NRC.]

2aPP17. Stage analysis of auditory event perception: Posture in human gait. Richard E. Pastore, Matthew J. Solomon, Jeremy R. Gaston, Jesse D. Flint, and Melody S. Berens (Dept. of Psych., Binghamton Univ., Binghamton, NY 13902-6000, pastore@Binghamton.edu)

One ultimate goal of auditory research is to understand how we perceive natural events. In our conceptualization, there are three component stages to auditory event perception: the source event, the sounds produced, and perception of the event. The current study carefully examines each of these three components, and the inter-relationships among their properties. The auditory event class for the current study is human gait in a fixed environment (e.g., shoes, walking surface, room), with posture (upright versus stooped) manipulated within individual walkers. We analyze the source events in terms of both the anthropometrical and dynamic biomechanical properties of the individual walker, and sound in terms of temporal and spectral properties of sole and heel collisions with the floor. Finally, perception is evaluated in terms of accuracy in identifying source properties (e.g., posture). In addition to statistical analyses of source-sound, sound-perception, and source-perception relationships, the source-sound link is investigated using biomechanical modeling. These analyses provide a comprehensive picture of the relationships among our three conceptual stages as well as identifying what is, and is not, relevant to perception of this specific source event. [Research supported by NSF.]

2aPP18. Temporal streaming based on fine structure cues. Nicolas Grimault (UMR CNRS 5020 Universit Lyon, 1 50 av. T Garnier, 69366, Lyon Cedex 07, France, nicolas.grimault@olfac.univ-lyon1.fr)

Alternating sequence of two *A* and *B* bursts of sound tends to be perceptually organized in different auditory streams when introducing either a spectral or a temporal difference between *A* and *B*. In general, the channeling theory of streaming predicts that any salient difference between the excitation patterns evoked by *A* and *B* sounds would lead to a segregated percept. Some previous studies have, however, evidenced that a sequence of sounds with similar spectral properties but with different temporal properties can be heard as segregated. In particular, temporal cues are probably responsible for the segregated percept when hearing a se-

quence of bursts of noises that are amplitude-modulated at different rates. All previous streaming experiments involved stimuli that had frequency components below 5000 Hz. As a consequence, the individual contribution of temporal fine structure cues and envelope cues could not be dissociated and remains largely undetermined. The current experiment is dedicated to test further the relative importance of both envelope and fine structure cues to segregate sequences of unresolved complex tones with different fundamental frequencies. Two high-frequency regions above and below 5000 Hz and several phase relationships leading to several temporal peak factors have been used in a subjective streaming task.

2aPP19. Mechanisms underlying the auditory continuity illusion. Daniel Pressnitzer, Julien Tardieu (CNRS UMR 9912, Ircam-CNRS, 1 place Stravinsky, F-75004 Paris, France), Richard Ragot, and Sylvain Baillet (CHU Pitié Salpêtrière, F-75013 Paris, France)

This study investigates the auditory continuity illusion, combining psychophysics and magnetoencephalography. Stimuli consisted of amplitude-modulated (AM) noise interrupted by bursts of louder, unmodulated noise. Subjective judgments confirmed that the AM was perceived as continuous, a case of illusory continuity. Psychophysical measurements showed that the illusory modulation had little effect on the detection of a physical modulation, i.e., the illusory modulation produced no modulation masking. Duration discrimination thresholds for the AM noise segments, however, were elevated by the illusion. A whole-head magnetoencephalographic system was used to record brain activity when listeners attended passively to the stimuli. The AM noise produced a modulated magnetic activity, the auditory steady-state response. The illusory modulation did not produce such a response, instead, a possible neural correlate of the illusion was found in transient evoked responses. When the AM was interrupted by silence, oscillatory activity in the gamma-band range as well as slow evoked potentials were observed at each AM onset. In the case of the illusion, these neural responses were largely reduced. Both sets of results are inconsistent with a restoration of the modulation in the case of illusory continuity. Rather, they point to a role for onset-detection mechanisms in auditory scene analysis.

TUESDAY MORNING, 25 MAY 2004

LIBERTY 1/2, 9:00 A.M. TO 12:00 NOON

Session 2aSA

Structural Acoustics and Vibration: Structural Vibration

Dean Capone, Chair

Flow and Structural Acoustics, Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804

Contributed Papers

9:00

2aSA1. Distributed versus concentrated load distribution for a shell subject to a rotating constant load. Mauro Pierucci (Dept. of Aersp. Eng., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-1308, mpierucci@engineering.sdsu.edu) and Scott Rosen (San Diego State Univ., San Diego, CA 92182)

When a beam or a shell is forced by a concentrated load at a discrete frequency, the structure response is composed of the sum of its natural modes. The concentrated load is always approximated by a point load as represented by the Dirac delta function. If the point load is replaced by a constant magnitude-distributed load with the same integrated value, the importance of the different modes is a function of the width of the loading

function. Results will be presented for a beam and a shell. The beam is driven by a stationary load at a constant frequency. The shell is forced by a time-invariant load rotating around the shell at a frequency.

9:15

2aSA2. Dynamic response of *in-vacuo* elliptic cylindrical shells. Jeffrey E. Boisvert (Naval Undersea Warfare Ctr., Newport, RI 02841) and Sabih I. Hayek (Penn State Univ., University Park, PA 16802, boisvertje@npt.nuwc.navy.mil)

The equations of motion for the vibration of elliptic cylindrical shells of constant thickness were derived using a Galerkin approach. The elastic strain energy density used in this derivation has seven independent kine-

matic variables: three displacements, two thickness shear, and two thickness stretch. The resulting seven coupled algebraic equations are symmetric and positive definite. The shell has a constant thickness, h , finite length, L , and is simply supported at its ends ($z=0,L$), where z is the axial coordinate. The elliptic cross section is defined by the shape parameter, a , and the half-length of the major axis, l . The modal solutions are expanded in a doubly infinite series of comparison functions in terms of circular functions in the angular and axial coordinates. Damping is introduced into the shell via a complex Young's modulus. Numerical results for the drive and transfer mobilities due to surface force excitations were obtained for several h/l and L/l ratios, and various shape parameters, including the limiting case of a simply supported cylindrical shell ($a \sim 100$). Sample mode shapes were obtained at selected resonant frequencies. [Work supported by ONR and the Navy/ASEE Summer Faculty Program.]

9:30

2aSA3. Predicting the surface wall pressure frequency spectra of submerged cylindrical bodies. Y. F. Hwang, W. K. Bonness, and S. A. Hambric (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804-0030)

This paper addresses the modeling of structural excitations caused by low speed turbulent boundary layer flows. When the Corcos-type cross-spectrum or its corresponding wavevector-frequency spectrum is used as the forcing function in a homogeneous turbulent flow, the point frequency spectrum is the common multiplier to the factors representing either the cross-spectral terms or the wavevector terms. The capability of accurately predicting the point frequency spectrum is the premise of obtaining an accurate cross-spectrum or wavevector-frequency spectrum. Several point-frequency spectral models have been evaluated against the measured data obtained from buoyancy propelled cylindrical bodies. It was found that predictions using the most recently published point-frequency spectral models by Smolyakov [Acoust. Phys. **46**(3), 342–347; translated from Akusti. Zh. **46**(3), 410–407 (2000)] and Goody [AIAA-2002-2565, 8th AIAA/CEAS Aeroacoustics Conference and Exhibit, Breckenridge, CO, (2002)] provide a reasonably good agreement with the measured data. [Work supported by ONR.]

9:45

2aSA4. A theory for thermoelastic damping in MEMS and NEMS flexural oscillators. Andrew Norris (Mech. and Aersp. Eng., Rutgers Univ., Piscataway, NJ 08854-8058, norris@rutgers.edu)

Intrinsic damping is a major roadblock in the drive to smaller high frequency RF oscillators, from the current mm size to MEMS and ultimately NEMS. Recent measurements on silicon MEMS platelike oscillators show that thermoelastic (TE) damping is the limiting damping mechanism. This talk will describe how the classical theory of Zener for TE damping needs to be revised for small thickness NEMS oscillators. A critical plate thickness is identified, separating thermally thick and thin regimes. In thermally thick oscillators the thermal diffusion is across the thickness, whereas in-plane thermal diffusion can dominate in thermally thin plates. Using the Kirchhoff assumption for the elastic deformation, it is shown that the local thermal relaxation loss depends upon the local state of vibrating flexure, specifically, the principal curvatures at a given point on the plate. The thermal loss is zero at points where the principal curvatures are equal and opposite, i.e., saddle shaped deformation. Effective plate equations are derived that incorporate TE loss as a damping term. A new form of the effective damping is obtained for both thermally thick and thin plates, generalizing the single relaxation Zener model. Applications to MEMS and NEMS oscillators will be discussed.

10:00

2aSA5. Coupled dynamic systems and Le Chatelier's principle in noise control. G. Maidanik and K. J. Becker (Code 7030, Carderock Div., Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817)

Investigation of coupling an externally driven dynamic system—a master dynamic system—to a passive one—an adjunct dynamic system—reveals that the response of the adjunct dynamic system affects the pre-coupled response of the master dynamic system. The responses, in the two dynamic systems when coupled, are estimated by the stored energies (E_s) and (E_0), respectively. Since the adjunct dynamic system, prior to coupling, was with zero (0) stored energy, $E_s^0=0$, the precoupled stored energy (E_0^0) in the master dynamic system is expected to be reduced to (E_0) when coupling is instituted; i.e., one expects $E_0 < E_0^0$. In this case a beneficial noise control of the master dynamic system would result from the coupling. It is argued that the change in the disposition of the stored energies as just described may not be the only change. The coupling may influence the external input power into the master dynamic system which may interfere with the expected noise control. Indeed, the coupling may influence the external input power such that the expected beneficial noise control may not materialize. Examples of these kinds of noise control reversals are cited.

10:15–10:30 Break

10:30

2aSA6. On the ensemble variance of the energy of random plates loaded by point forces and distributed pressure fields. Vincent Cotoni, Bryce Gardner (ESI R&D, 12555 High Bluff Dr., Ste. 250, San Diego, CA 92130, vcotoni@vasci.com), and Robin Langley (Univ. of Cambridge, Cambridge CB2 1PZ, UK)

Statistical energy analysis (SEA) is used to predict the dynamic behavior of structural-acoustic systems at high frequencies. SEA predicts the mean vibrational energy over an ensemble of nominally equivalent systems. For example, a collection of individual cars coming off an assembly line would make an ensemble whose mean response is predicted by SEA. Recently a prediction of the variance of the energy in the SEA ensemble has been developed. At high frequencies where small changes in the system cause shifts in natural frequency that are larger than the average spacing of the natural frequencies, the variance becomes independent of the detailed nature of the differences between the members of the ensemble. This variance of energy can then be predicted with information available in an SEA model and with some few additional parameters describing the nature of the excitations and couplings. This new formulation is applied to an ensemble of random plates loaded by point forces and distributed pressure fields. An accurate prediction of the observed variance is demonstrated.

10:45

2aSA7. The influence of stiffened plate modifications on the energy balance. Marek Iwaniec and Ryszard Panuszka (Univ. of Sci. and Technol., Structural Acoust. and Biomed. Eng. Lab., Al. Mickiewicza 30, 30-059 Krakow, Poland)

The change in plate stiffness introduced by a system of stiffeners slightly changes the whole system mass and results in a significant change in natural frequencies, distribution and values of vibrational energy. Stiffeners can be treated as an additional subsystem that not only changes the plate stiffness but also accumulates and dissipates a part of vibrational energy. The energy balance analysis of a stiffened plate was carried out for a plate treated as an assembly of a homogeneous plate and a system of beams. On the basis of modal participation factors the energies dissipated in plates and system of stiffeners were determined. The natural frequencies and modal participation factors were computed as a solution of an unconstrained optimization problem by the use of the Nelder Mead optimization algorithm and compared to a twice faster genetic algorithm. Theoretical computations of a homogeneous plate and stiffeners energies were verified

by energy distribution carried out on the basis of frequency response functions of FEM models. The influence of geometrical and material properties on natural frequencies and energy dissipated by plate and stiffeners was determined.

11:00

2aSA8. Dynamics of transversely isotropic cylinders. Subrata K. Bhattacharyya and Chiruvai P. Vendhan (Ocean Eng. Dept., I.I.T. Madras, Chennai, Tamilnadu 600036, India, vendhan@iitm.ac.in)

Studies on wave propagation in cylinders have a long history [M. J. Berliner and R. Solecki, *J. Acoust. Soc. Am.* **99**, 1841–1853 (1996)]. Onoe *et al.* [*J. Appl. Mech.* **29**, 729–734 (1962)] studied the attenuating and propagating modes of an isotropic cylindrical rod. Mirsky [*J. Acoust. Soc. Am.* **37**, 106–1021 (1965)] presented the solution for harmonic wave propagation in a transversely isotropic (TI) cylinder in vacuo. Sinha *et al.* [*J. Acoust. Soc. Am.* **91**, 1132–1143 (1992)] studied isotropic cylinders in contact with fluids employing the complex wave number solution. Bhattacharyya [Ph.D. thesis, Dept. of Ocean Eng., I.I.T. Madras (1989)] generalized Mirsky's work to study a TI cylinder in contact with fluids. Berliner and Solecki (cited above) also presented a similar solution and applied it to strongly TI cylinders. The present article is concerned with the vibration of a TI cylinder under prescribed harmonic end loading. The problem may be viewed as the dynamic analog of the Saint-Venant problem. The forced vibration solution is facilitated by an orthogonality relation among the radial eigenfunctions corresponding to the real and complex roots of the dispersion relation [W. B. Fraser, *J. Sound Vib.* **43**, 568–571 (1975)]. This approach can be used for the rigorous solution of finite cylinders.

11:15

2aSA9. High-frequency vibrations of a piezomagnetic plate. Gulay Altay (Dept. of Civil Eng., Faculty of Eng., Bogazici Univ., Bebek, 34342 Istanbul, Turkey, askarg@boun.edu.tr) and M. Cengiz Dokmeci (Istanbul Tech. Univ., Gummussuyu, 34430 Istanbul, Turkey)

This paper deals with a consistent derivation of the hierarchic system of two-dimensional approximate equations of a piezomagnetic plate at high frequency where the wavelength is of the order of magnitude or smaller than the plate thickness. To begin with, a generalized variational principle is reported for piezomagnetism, and the field variables are represented by the power series expansions in the thickness coordinate. Next, with the aid of the variational principle together with the series expansions, the system of plate equations is derived in invariant, differential and fully variational forms. The system of equations that may be readily expressed in a particular coordinate system most appropriate to the plate geometry is capable of predicting all the types of vibrations at both low and high frequency. Also, the uniqueness is investigated in solutions of the system of plate equations, and the conditions are enumerated for the uniqueness. Further, certain cases involving special geometry, material

properties and types of vibrations are indicated. The resulting equations agree with and generalize some of earlier plate equations [cf. the authors, *Int. J. Solids Struct.* **40**, 4699–4706 (2003)]. [Work supported by TUBA.]

11:30

2aSA10. Free flexural vibrations of super elliptic plates. Murat Altekin (Faculty of Civil Eng., Yildiz Tech. Univ., Besiktas, 34330 Istanbul, Turkey, altekin@yildiz.edu.tr), Gulay Altay (Bogazici Univ., 34342 Istanbul, Turkey), and M. Cengiz Dokmeci (Istanbul Tech. Univ., 34430 Istanbul, Turkey)

This paper deals with a direct problem of the free flexural vibrations of a plate with rounded corners on the basis of the theory of thin plates. The plate of uniform thickness is made of linear elastic, isotropic materials, and its periphery is given by a super elliptic function with a power. The super elliptic power defines the shape of the plate ranging from an ellipse to a rectangle and indicates the degree of roundness. The method of solution is based on the method of separation of variables together with certain methods of weighted residuals (i.e., the method of moments, and the Galerkin and least squares methods). The shape functions are chosen in the form of double series polynomials, and they satisfy either a simply supported or a clamped boundary condition. Some numerical results are reported for the vibration frequencies in the case of both symmetric and antisymmetric mode shapes of the plate. The results are compared with the existing literature, and the convergence of solutions is discussed. [Work supported in part by TUBA.]

11:45

2aSA11. Hamilton's principle applied to piezomagnetism and related variational principles. M. Cengiz Dokmeci (Istanbul Tech. Univ., P.K.9, Gumussuyu, 34430 Istanbul, Turkey, cengiz.dokmeci@itu.edu.tr) and Gulay Altay (Bogazici Univ., Bebek, 34342 Istanbul, Turkey)

In piezomagnetism, the fundamental equations have been developed in differential form [e.g., V. I. Alshits and A. N. Darinskii, *Wave Motion* **15**, 265–283 (1992)]. Alternatively, they may be expressed in variational form with its well-known features; this is the topic of this paper. First, the magnetic vector, that is, the gradient of the magnetic potential, is introduced [cf. the authors, *Int. J. Solids Struct.* **40**, 4699–4706 (2003)]. Second, the sufficient conditions based on the energy argument are enumerated for a unique solution in the fundamental equations. Third, Hamilton's principle is stated and a three-field variational principle is obtained. The principle yields only the divergence equations and some natural boundary conditions, and it has the remaining fundamental equations as its constraint conditions. The conditions are generally undesirable in computation, and they are accordingly removed through an involutory transformation [e.g., the authors, *Int. J. Eng. Sci.* **40**, 457–489 (2002)]. Thus, a unified variational principle operating on all the field variables is derived in piezomagnetism. The principle is shown, as special cases, to recover some of earlier ones. [Work supported by TUBA.]

Session 2aSC**Speech Communication: Forty Years of VOT (Voice Onset Time) (Lecture/Poster Session)**

Anders Lofqvist, Cochair

Department of Logopedics and Phoniatrics, Lasarettet, Lund S-221 85, Sweden

Laura Koenig, Cochair

*Haskins Laboratories, 270 Crown Street, New Haven, Connecticut 06511***Invited Papers****8:00****2aSC1. Lisker and Abramson: Teaching researchers.** Lawrence J. Raphael (Dept. of Commun. Sci. and Disord., Adelphi Univ., Garden City, NY 11530, raphael@adelphi.edu)

The “discovery” of voice onset time (VOT) has, perhaps, stimulated more research projects than any other comparable measure. When the fathers of VOT, Leigh Lisker and Arthur Abramson, published their paper in 1964, other researchers were quick to recognize the implications of their findings and of their methodology. Those in the neighborhood of the creation in the mid-1960s were carefully trained in the proper procedures for making VOT measurements and were taught when such measures were, or were not, appropriate. In this regard, the dynamic duo was simply sustaining the tradition of using research as the basis of teaching, a tradition which both Arthur and Leigh have continued to the present. For those of us who have been fortunate enough to associate with them, their legacies as teachers stand as high in their list of accomplishments as anything else. Any academic researcher would do well to emulate their effective and often unique teaching methods, some of which will be described in this paper.

8:20**2aSC2. Producing VOT contrasts.** Anders Lofqvist (Haskins Labs., 270 Crown St., New Haven, CT 06511, lofquist@haskins.yale.edu)

The development of voice onset time (VOT) as an acoustic index for studying and classifying stop consonants also prompted a large number of studies examining laryngeal activity and interarticulator timing related to VOT. A collaboration between the Research Institute of Logopedics and Phoniatrics at the University of Tokyo and Haskins Laboratories resulted in a long line of studies using electromyographic and other techniques that provided much of the empirical foundations for what we know about laryngeal function in speech, in particular the production of voiced and voiceless consonants. This presentation will review the articulatory control of VOT differences. To make a consonant voiceless, a speaker uses a combination of glottal abduction and vocal fold tensing. The distinction between voiceless stops with long and short VOT is basically due to a difference in the timing between the glottal abduction gesture and the oral closing and opening gestures. Variations in the size of the glottal gesture also occur. More generally, variations in interarticulator timing between glottal and oral movements are used to produce the different stop categories that occur in the languages of the world. [Work supported by NIH.]

8:40**2aSC3. VOT and the perception of voicing.** Robert E. Remez (Dept. of Psych., Barnard College, 3009 Broadway, New York, NY 10027, remez@columbia.edu)

In explaining the ability to distinguish phonemes, linguists have described the dimension of voicing. Acoustic analyses have identified many correlates of the voicing contrast in initial, medial, and final consonants within syllables, and these in turn have motivated studies of the perceptual resolution of voicing. The framing conceptualization articulated by Lisker and Abramson 40 years ago in physiological, phonetic, and perceptual studies has been widely influential, and research on voicing now adopts their perspective without reservation. Their original survey included languages with two voicing categories (Dutch, Puerto Rican Spanish, Hungarian, Tamil, Cantonese, English), three voicing categories (Eastern Armenian, Thai, Korean), and four voicing categories (Hindi, Marathi). Perceptual studies inspired by this work have also ranged widely, including tests with different languages and with listeners of several species. The profound value of the analyses of Lisker and Abramson is evident in the empirical traction provided by the concept of VOT in research on the every important perceptual question about speech and language in our era. Some of these classic perceptual investigations will be reviewed. [Research supported by NIH (DC00308).]

9:00**2aSC4. VOT in cross language and comparative phonetics.** Ian Maddieson (Dept. of Linguist., Univ. of California, 1203 Dwinelle Hall #2650, Berkeley, CA 94720-2650, ianm@socrates.Berkeley.edu)

Since the earliest introduction of the concept of voice onset time to describe laryngeal timing in stop consonants (Lisker and Abramson, 1964), this measure has been used as a tool for discussing phonetic similarities and differences among languages. The distinction between broad categories of leading, aligned, or lagging voice onset in relation to oral release is useful across many languages. There are also more subtle effects on VOT duration, such as some concerning major consonant places of articulation and

following vowel contexts which have proved to be repeated across languages. Combining these regularities allows relatively precise prediction of VOT in timing models. The VOT concept has also been extended to discussion of certain less-common classes of segments such as ejective stops, as well as to fricatives and affricates. The latter case raises interesting questions of distinguishing between frication noise and aspiration noise, and of what to take as the consonant's release. How best to characterize laryngeal timing around the release of segments not followed by a vowel or voiced sonorant will also be considered.

9:20

2aSC5. VOT in speech-disordered individuals: History, theory, data, reminiscence. Gary Weismer (Dept. of Communicative Disord. and Waisman Ctr., Univ. of Wisconsin—Madison, 1975 Willow Dr., Madison, WI 53706)

Forty years ago Lisker and Abramson published their landmark paper on VOT; the speech-research world has never been the same. The concept of VOT as a measure relevant to phonology, speech physiology, and speech perception made it a prime choice for scientists who saw an opportunity to exploit the techniques and analytic frameworks of "speech science" in the study of speech disorders. Modifications of VOT in speech disorders have been used to draw specific inferences concerning phonological representations, glottal-supraglottal timing, and speech intelligibility. This presentation will provide a review of work on VOT in speech disorders, including (among others) stuttering, hearing impairment, and neurogenic disorders. An attempt will be made to collect published data in summary graphic form, and to discuss their implications. Emphasis will be placed on how VOT has been used to inform theories of disordered speech production. I will close with some personal comments about the influence (unbeknownst to them) these two outstanding scientists had on me in the 1970s, when under the spell of their work I first became aware that the world of speech research did not start and end with moving parts.

9:40

2aSC6. VOT and hearing impairment. Harlan Lane and Joseph Perkell (Res. Lab. of Electron., MIT, Rm. 36-511, 50 Vassar St., Cambridge, MA 02139, lane@speech.mit.edu)

When deafened adults recover some hearing after receiving a cochlear implant, numerous changes in their speech occur at both phonemic and suprasegmental levels. If a change toward normative values is observed for some phonemic parameter, it may be attributed to the restored hearing; however, it may be a by-product of a suprasegmental change. Consistent with results reported for speakers with normal hearing, Lane *et al.* [J. Acoust. Soc. Am. **98**, 3096–3106 (1995)] observed in implant users that VOT varies approximately linearly with syllable duration. Therefore, in comparing pre- and postimplant measures of VOT in five speakers, each token's VOT was adjusted for the change in syllable duration of that token relative to the mean syllable duration in a baseline session (called VOTc). Preimplant, the deaf speakers characteristically uttered plosives with abnormally short VOTc. With some hearing restored, four of the five lengthened VOTc. Changes in voiced plosives' VOTc with restored hearing were correlated with changes in SPL. Some of the reliable VOTc increases that were not correlated with SPL may have been caused by auditory validation of an internal model for phoneme production. Recent studies of VOT in hearing-impaired speakers will be reviewed in this light. [Work supported by NIDCD, NIH.]

10:00–10:15 Break

Poster Papers

All posters will be on display and all contributors will be at their posters from 10:15 a.m. to 12:00 noon.

2aSC7. A comparison of English versus Spanish voicing perception using natural speech. Joan M. Sinnott and Jazmin Camchong (Psych. Dept., Univ. of South Alabama, Mobile, AL 36688)

English versus Spanish voicing perception over the past 30 years has been almost exclusively studied using synthetic VOT continua, and there has been very little research using natural VOT stimuli. This study used a balanced symmetrical design to explore the effects of training English and Spanish listeners to categorize natural tokens of English versus Spanish /b-p/ using four different vowels /i,e,a,u/. Extensive training with feedback was conducted over several months, and percent correct categorization and reaction time were analyzed. Results showed that each language group consistently exhibited enhanced performance for native speech, and this difference persisted with training. For example, reaction times leveled off at approximately 50 ms faster for native versus non-native speech. It was concluded that, while lab training can improve the ability to perceive a non-native voicing contrast, it does not result in native-like perception. Some preliminary results from monkeys using the same stimuli and procedure indicate that, unlike human adults, monkeys are more like human infants and find English and Spanish voicing contrasts equally salient. [Research supported by NIH.]

2aSC8. Voice onset time and global foreign accent in mature Japanese–English bilinguals. Timothy J. Riney (Grad. School Div. of Education, Intl. Christian Univ., Tokyo, Japan)

Experiment 1 investigates to what degree the phonetic category established in childhood for an L1 sound may evolve gradually [Flege, Speech Learning Model (1995)]. VOT is assessed in two groups of five bilingual (English and Japanese) speakers, one L1 English and the other L1 Japanese, ages 35–60 with little or no L2 exposure before age 15, but working in the L2 regularly for 15–35 years in adulthood. Both groups read lists of English and Japanese words containing word initial /p/, /t/, and /k/; 880 tokens (10 speakers × 2 languages × 44 tokens per language) were examined. The findings for both groups were that (a) the L1 VOT values did not change much over time and (b) the L2 VOT values were halfway between the L1 and target language values. Experiment 2 investigates the relationship between individual VOT and accent. The ten speakers were asked to read five Japanese sentences, now digitized. Ten additional L1 Japanese participants will rate these sentences for accent in March 2004. The prediction is that the American speakers who have the most Japanese accents will also have VOT values that are closest to L1 Japanese values.

2aSC9. The acquisition of voice onset time by English-learning infants. Joanna H. Lowenstein and Susan Nittrouer (Ctr. for Persons with Disabilities, Utah State Univ., 6800 Old Main Hill, Logan, UT 84322)

Lisker and Abramson [Word 20, 384-422 (1964)] showed that voice onset time (VOT) for word-initial stops labeled as voiced and voiceless differ across languages, thus demonstrating that VOT is learned. Over the years, several studies have examined the developmental course of acquiring VOT for several languages, but questions remain regarding the time course of this development. Earlier studies of VOT acquisition have demonstrated that English learning infants initially produce stops with short-lag VOTs only, then go through a stage of ambiguous VOTs, finally developing two mature categories of voiced and voiceless. This study analyzed mean VOT and variability in word-initial stops produced in spontaneous samples by three infants, taped at approximately 2-month intervals between the ages of 12 months and 30 months. As predicted, these infants produced primarily voiced stops ($VOT < 40$ ms) with moderate variability until approximately 21 months. Then longer lag stops began to appear and variability increased. Outliers (both voiced stops with extreme prevoicing and very long lag voiceless stops) were found. By the last recording session, all three infants showed mean voiced and voiceless VOTs approaching adult values, but still exhibited greater variability. [Work supported by NIDCD Grant No. DC-00633.]

2aSC10. Voice-onset time and buzz-onset time identification: A ROC analysis. Luis E. Lopez-Bascuas (Universidad Complutense de Madrid, Facultad de Psicología, Campus de Somosaguas, 28223 Madrid, Spain, lelopezb@psi.ucm.es), Burton S. Rosner (Univ. of Oxford, Oxford OX1 2JF, England), and Jose E. Garcia-Albea (Universitat Rovira I Virgili, 43007 Tarragona, Spain)

Previous studies have employed signal detection theory to analyze data from speech and nonspeech experiments. Typically, signal distributions were assumed to be Gaussian. Schouten and van Hensen [J. Acoust. Soc. Am. 104, 2980-2990 (1998)] explicitly tested this assumption for an intensity continuum and a speech continuum. They measured response distributions directly and, assuming an interval scale, concluded that the Gaussian assumption held for both continua. However, Pastore and Macmillan [J. Acoust. Soc. Am. 111, 2432 (2002)] applied ROC analysis to Schouten and van Hensen's data, assuming only an ordinal scale. Their ROC curves supported the Gaussian assumption for the nonspeech signals only. Previously, Lopez-Bascuas [Proc. Audit. Bas. Speech Percept., 158-161 (1997)] found evidence with a rating scale procedure that the Gaussian model was inadequate for a voice-onset time continuum but not for a noise-buzz continuum. Both continua contained ten stimuli with asynchronies ranging from -35 ms to +55 ms. ROC curves (double-probability plots) are now reported for each pair of adjacent stimuli on the two continua. Both speech and nonspeech ROCs often appeared nonlinear, indicating non-Gaussian signal distributions under the usual zero-variance assumption for response criteria.

2aSC11. The role of formant transitions in the perception of syllable final voicing. W. J. Warren (Dept. of Linguist., Univ. of Texas, 1 University Station B5100, Austin, TX 78712-0198) and A. E. Coren (Univ. of Texas, Austin, TX 78712-0187)

This experiment examines the role of formant transitions as a cue in syllable final voicing distinctions, specifically between low vowels and alveolar stops. Past studies have shown that when token length is a variable, listeners identify shorter tokens with voiceless codas and longer tokens with longer tokens. Ten tokens of 200 ms were synthesized from an original token and randomly played to 15 listeners. All tokens were derived from an original with steady state and formant transition period of the vowel at 100 ms each. F1 cutoff was 250 Hz and F2 cutoff was 1600 Hz. To control for token time, as transitional period was taken off in intervals of 5 ms it was then added to the steady state so that a length of 200 ms could be preserved for all tokens. Previous experiments have shown that adding 100 ms of steady state vowel onto tokens of 350 ms

produces no shift in voicing distinctions, and the results here strengthen that finding. It has been shown that listeners can quickly learn to associate formant transitions with syllable final voicing distinctions when the length of the tokens is controlled.

2aSC12. Phonetic specifications in infant-directed speech. Ulla Sundberg (Dept. of Linguist., Stockholm Univ., SE-10691 Stockholm, Sweden, ulla@ling.su.se)

In contrast to the overspecification of prosodic aspects and the expansion of the vowel space typically observed in infant-directed speech (IDS), VOT measurements of voiceless stop consonants occurring in Swedish mothers' speech to their 3-month-old infants revealed surprisingly lower values in IDS than in the mothers' adult-direct speech (ADS). Since the mothers' underspecification of consonantal contrasts in early IDS, while at the same time exaggerating prosodic and vocal contrasts, may indicate an adjustment to the phonetic aspects that they perceive to be adequate to the infant's developmental level, the present study of VOT in IDS towards 12-month-olds was carried out in order to determine the age interval within which mothers may start to adopt adult-like VOT contrasts in their IDS. The VOT measurements from the IDS aimed at their 12-month-olds showed instead significantly longer durations than in ADS. It is hypothesized that this age-dependent use of consonantal specifications in IDS may reflect the mothers' adjustments to their perception of the infants' communicative capabilities, from dominating phatic function, conveyed by deeply modulated speech and exaggerated vowel contrast in early IDS, towards a focus on lexical content by the end of the first year of life.

2aSC13. Control of short lag VOT (voice-onset time) for voiced English stops. Jeff Berry (Univ. of Wisconsin—Madison, Waisman Ctr., Rm. 493, 1500 Highland Ave., Madison, WI 53705, jberry@wisc.edu)

The American English stops /b/, /d/, and /g/ are produced with negative VOT (-150-0 ms) or short-lag VOT (0-30 ms). These two variants have been shown to occur with approximately equal frequency across speakers, although individual preferences are common. It has been hypothesized that the characteristic VOTs of short-lag stops are determined by relatively simpler articulatory control mechanisms than negative VOT variants. Specifically, if the vocal folds are approximated prior to constriction release, and intraoral pressure is vented only at the point of maximum constriction (without supraglottal expansion), the VOT will be determined by the venting process controlled by the time-varying constriction aperture. Thus, the release kinematics could account for effects of place of articulation and vowel context on short-lag VOTs. In the current work, the plausibility of this hypothesis is evaluated. The kinematics of constriction release gestures was examined from x-ray microbeam data. Descriptions of the time-varying geometries of constriction releases were input to a simple model of breath-stream dynamics and simulated VOT values were obtained. The descriptive and simulated data are consistent with the hypothesis that the characteristic durations of short-lag VOTs are determined by the venting process controlled by the supraglottal stop release gesture.

2aSC14. Perception of Spanish, Parkinsonian speech by English listeners. Michael Fraas (Dept. of Commun. Sci. and Disord., Univ. of New Hampshire, Durham, NH 03824, michael.fraas@unh.edu) and Suzanne Boyce (Univ. of Cincinnati, Cincinnati, OH 45202)

It is well established that when two languages differ in voicing patterns for obstruents, listeners are likely to misidentify obstruents in a non-native language according to the patterns of their native language. It has also been established that the normal patterns that distinguish voiced and voiceless stops in English are disrupted in speakers with Parkinson's disease. Accordingly, English-speaking clinicians not familiar with the voicing characteristics of another language, e.g. Spanish, run the risk of erroneously interpreting the utterances of clients who speak that language as natives. In a study of the voice-onset time characteristics of stops spoken

by Spanish speakers with Parkinson's disease and a matched set of control Spanish speakers (Fraas, 2002; Pan-American/Iberian Meeting on Acoustics), we determined that Spanish speakers with PD produce shorter VOTs for voiced stops and longer VOTs for voiceless stops as compared with Spanish-speaking controls. These longer VOTs were judged to be appropriate by English listeners, although they are not characteristic of normal Spanish production. These results imply that English-speaking clinicians are likely to err in not identifying voicing as a target of remediation for Spanish speakers with PD.

2aSC15. Contrastive voicing acquisition in 2-year-old children: Preliminary data. Elaine R. Hitchcock (New York Univ., New York, NY, hitchcocke@mail.montclair.edu) and Laura L. Koenig (Haskins Labs., CT 06511)

Earlier studies using voice-onset time (VOT) as the acoustic marker of contrastive voicing acquisition in English have differed widely in method and statistical procedures. Research in this area has shown three primary patterns of voicing acquisition for English stop consonants. One report indicates children demonstrate discrete voicing categories with adult-like VOT values as early as 2 years old. Other reports suggest a subperceptual distinction of the voicing contrast, followed by an exaggerated voicing contrast with evidence of more adult-like discrete categories by 3 years old. Still other reports suggest no distinction between the voicing categories until approximately 3 years. This work investigates voicing acquisition in three typically developing, English-speaking 2-year-old children. The subjects were recorded every 2 weeks for 4–6 months. Approximately 15–20 tokens were elicited for four target utterances containing initial /b p t d/. Frequency distribution, measures of central tendency, and skewness will be calculated for every recording session of each child. Discussion will focus on the development of contrastive VOT categories and their stability over time. These data will contribute to our understanding of laryngeal timing for English stop consonants in young children.

2aSC16. Physiological evidence for a temporal processing mechanism underlying voice-onset time (VOT) encoding. Mitchell Steinschneider, Yonatan I. Fishman (Dept. of Neurology, Kennedy Ctr., Rm. 322, Albert Einstein College of Medicine, 1300 Morris Park Ave., Bronx, NY 10461, steinsch@aecom.yu.edu), Igor O. Volkov, and Matthew A. HowardIII (Univ. of Iowa College of Medicine, Iowa City, IA 52242)

Despite decades of psychoacoustical research, the detailed neural mechanisms underlying VOT encoding remain obscure. Evidence collected from direct recordings in auditory cortex of human subjects undergoing surgical evaluation for medically intractable epilepsy, and from primary auditory cortex in monkeys, supports a temporal processing mechanism as a principal means by which VOT is encoded by the brain. This mechanism, as proposed by Pisoni [J. Acoust. Soc. Am. **77**, 1352–1361 (1977)], argues that the perceptual discrimination of voiced from unvoiced stop consonants is based, in part, on whether consonant release and voicing onset are perceived as occurring simultaneously or sequentially. Neural activity in auditory cortex offers physiologically plausible parallels to this perceptual scheme that can help account for the distribu-

tion of typical VOT values used by the majority of the world's languages, categorical perception of VOT, and perceptual boundary shifts that occur with changes in stop-consonant place of articulation and when nonspeech analogs of VOT are used. These responses in primary auditory cortex are poised to provide powerful inputs to later processing areas, where they can be integrated with other acoustical, visual, and language-related inputs known to modulate VOT perception. [Work supported by DC00657 and DC00120.]

2aSC17. Bilinguals' categorical perceptual shift when producing Spanish or English words: Electrophysiological correlates. Adrian Garcia-Sierra and Craig Champlin (Univ. of Texas, 1 University Station A1100, Austin, TX 78712)

In a previous study Mexican–American bilinguals identified synthetic /ga/-/ka/ syllables varying in VOT in two language conditions (Spanish and English). The perceived voicing boundary of the bilinguals varied depending on the language they had produced before the test session. In the present study event-related potentials (ERPs) were recorded from bilinguals in both language conditions. A classic oddball paradigm was used to obtain mismatch negativity (MMN) responses as an indication of categorical perception. Participants were tested in two sessions one week apart with the language conditions counterbalanced. In each test session, the electrophysiological recording was paused every 5 min in order to avoid MMN habituation. During the pause, bilinguals produced a set of Spanish or English words to keep them focused in the language of interest. The results indicated smaller MMN amplitude in the language condition where deviant and standard stimuli were more likely to be perceived as part of the same category. [Work supported by the Department of Communication Sciences and Disorders Univ. of Texas—Austin.]

2aSC18. Perception of a non-native speech contrast: Voiced and voiceless stops as perceived by Tamil speakers. Sylwia Tur (Dept. of Linguist., Univ. of Washington, Seattle, WA 98195-4340)

The effect of linguistic experience plays a significant role in how speech sounds are perceived. The findings of many studies imply that the perception of non-native contrasts depends on their status in the native language of the listener. Tamil is a language with a single voicing category. All stop consonants in Tamil are phonemically voiceless, though allophonic voicing has been observed in spoken Tamil. The present study examined how native Tamil speakers and English controls perceived voiced and voiceless bilabial, alveolar, and velar stops in English. Voice onset time (VOT) was manipulated for editing of naturally produced stimuli with increasingly longer continuum. Perceptual data was collected from 16 Tamil and 16 English speakers. Experiment 1 was an AX task in which subjects responded same or different to 162 pairs of stimuli. Experiment 2 was a forced choice ID task in which subjects identified 99 individually presented stimuli as pa, ta, ka or ba, da, ga. Experiments show statistically significant differences between Tamil and English speakers in their perception of English stop consonants. Results of the study imply that the allophonic status of voiced stops in Tamil does not aid the Tamil speakers in perceiving phonemically voiced stops in English.

Session 2aSPa

Signal Processing in Acoustics and Underwater Acoustics: Acoustic Communications—Space/Time Coding, Modulation, Processing and Propagation Effects

James C. Preisig, Chair

Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 207, Woods Hole, Massachusetts 02543-1053

Chair's Introduction—8:00

Invited Papers

8:05

2aSPa1. Acoustic communication systems and space/time processing. Arthur B. Baggeroer (MIT, Cambridge, MA 02139)

Acoustic communication research for the last three decades has concentrated on point-to-point systems. Efforts are now evolving which exploit the spatial aspects of the underwater channel. First, there are systems with array for transmitters and/or receivers. Because the ocean randomizes the propagation, information can be received over many “paths;” for this beamforming and, more generally, singular value decomposition methods have been used to increase channel capacity. With multiple nodes in a distributed system, the problems of (i) time latency due to the relatively slow speed of sound and (ii) range, or multipath, spread become much more problematic; consequently, the space/time architecture of using time, frequency and space becomes very important and this differentiates underwater systems from cellular phone technology. The network structure, e.g., centralized or link connected leads to different signaling protocols and the potential for complicated interaction and interference. Nevertheless, many advances including both space/time diversity and coding are now popular research topics which will lead to more reliable and higher data rate acoustic communication systems.

8:25

2aSPa2. Optimization of transmit/receive array processing for high rate acoustic communications. Milica Stojanovic (MIT, Cambridge, MA 02139)

A broadband acoustic communication link between a user equipped with a single transmit/receive element and a station equipped with an array is considered. System optimization is conducted to obtain transmit/receive filtering techniques that provide best performance over a multipath channel under varying system constraints. Two classes of systems are considered: those that aim for complete intersymbol interference suppression and those that use equalization at the receiver side. Both systems are considered with or without the possibility to implement a perfect channel feedback. Transmit/receive filters are derived analytically, and performance of each scheme is assessed through its data detection SNR. Performance of various techniques is compared analytically, using an example of a shallow water channel, where transmission at high bit rate results in extensive delay spread.

Contributed Papers

8:45

2aSPa3. Simultaneously multiple-depth coherent communications using time reversal. Heechun Song, W. S. Hodgkiss, W. A. Kuperman, Philippe Roux, T. Akal (MPL/SIO, 9500 Gilman Dr., La Jolla, CA 92093-0238), and M. Stevenson (NATO SACLANT Undersea Res. Ctr., La Spezia 19138, Italy)

A recent time reversal experiment demonstrated that multiple foci can be projected from an array of sources to the same range but at different depths. This “multiple input, multiple output (MIMO)” process potentially can improve the information data rate. This paper presents the experimental results of binary phase shift keying (BPSK) communications at 3.5 kHz with a 500-Hz bandwidth where two different messages were sent simultaneously to different depths at 9-km range in 110-m-deep shallow water.

9:00

2aSPa4. Multichannel time reversal communications in a highly reverberative environment. James V. Candy, Brian Guidry, Andrew Poggio, and Claudia Kent (Univ. of California, Lawrence Livermore Natl. Lab., P.O. Box 808, L-156, Livermore, CA 94551, candy1@llnl.gov)

The development of point-to-point time-reversal (T/R) communications in a highly reverberative environment was discussed previously [Candy *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **114**, 2367]. This paper focuses on the extension of that effort to the multichannel case. Here, we discuss the theoretical development of a suite of multichannel TR receiver realizations, similar to the point-to-point case, using an acoustic T/R array and a set of client stations. The performance of these processors on both simulated and experimental data is discussed in detail. The experiment is provided by a stairwell between two floors of a noisy building. The stairwell is populated with obstructions (pipes, rails, wall, etc.) and a 90-deg landing—clearly a highly reverberative environment. It is shown that the

multichannel receivers perform quite well when compared to their point-to-point counterparts, and are able to reliably extract the transmitted code from the noisy measurements.

9:15

2aSPa5. Time-reversal communication through a highly reverberant medium. David H. Chambers, Claudia A. Kent, and Alan W. Meyer (Lawrence Livermore Natl. Lab., P.O. Box 808, Livermore, CA 94551)

An ultrasonic time-reversal array system is used to transmit communication signals across an aluminum slab with a large number of holes drilled through it. The hole pattern was designed to greatly attenuate the direct propagation path between each transmitter and receiver, thereby forcing the communication signal to be carried primarily by the random, scattered field. Four approaches to time-reversal communications [see Candy *et al.*, *J. Acoust. Soc. Am. Suppl.* 1 **114**, 2367 (2003)] were used to establish clear (negligible symbol error) communication channels through the slab. Both point-to-point and array-to-point configurations were tested. Comparisons between these approaches are shown using both simulated and experimental measurements. [Work performed under the auspices of the U. S. Department of Energy by the University of California, Lawrence Livermore National Laboratory under Contract No. W-7405-Eng-48.]

9:30

2aSPa6. Using acoustic orthogonal signals in shallow water time-reversal applications. Thomas Folegot (Atlantide, Technopole Brest Iroise, CS 23866, 29238 Brest, France), Philippe Roux, William A. Kuperman, William S. Hodgkiss, Hee Chun Song, Tuncay Akal (Scripps Inst. of Oceanogr., La Jolla, CA), and Mark Stevenson (NATO SACLANT Undersea Res. Ctr., La Spezia 19138, Italy)

Orthogonal broadband signals (adaptive instant record, AIR Signals) are constructed that are particularly suitable to shallow water multipath propagation. They have similarities with the multi-carrier code division multiple access (CDMA) methods which have arisen from mobile telephony, and allow communication between several sources and several receivers. The period of the orthogonal signals is related to the channel time spreading in order to provide an exact and simultaneous measure of the transfer function between all sources and receivers on the whole transducer frequency band. This method which optimizes the signal-to-noise ratio is relevant to fluctuating environments and particularly to time reversal applications. The first experimental demonstration at the ultrasonic scale has shown the efficiency of these signals in a stationary environment as well as their robustness to absorption and to bottom reverberation. This presentation describes the experimental use of these signals with a time reversal mirror in the Mediterranean Sea during an April 2003 experiment.

9:45

2aSPa7. Adaptive instant record signals applied to shallow water detection. Thomas Folegot (Laboratoire Ondes et Acoustique, Université Denis Diderot, UMR CNRS 7587, ESPCI, 10 rue Vauquelin, 75231 Paris Cedex 05, France and Atlantide, Marine Sci. and Technol. Dept., Technopôle Brest-Iroise, CS23866, 29238 Brest Cedex 3, France), Julien de Rosny, Claire Prada, and Mathias Fink (Université Denis Diderot, 75005 Paris, France)

Time reversal arrays are becoming common tools whether for detection, tomography or communication. These applications require the measurement of the response from the array to one or several receivers. The most natural way to record different impulse responses between several points is to generate pulses successively from each emitting point and directly record all the impulse responses on the recording points. However, this method is very time consuming and inefficient in terms of signal-to-noise ratio. Hence, in this work, we propose an original way of sending continuous signals simultaneously from all the sources, recording all the pressure fields on the receivers and processing them in order to extract the exact impulse responses by matched filter. To this end, the signals are adapted to the environment and, more specifically, to highly dispersive media. These adaptive instant records signals (AIRS) are used experimen-

tally to detect targets using the time reversal operator decomposition method. The quality of the 15×15 transfer functions acquired simultaneously, and therefore, the detection capability is demonstrated in shallow water in the presence of bottom absorption and reverberation. Finally, the connection of AIRS with CDMA and FDMA that are two coding techniques used in telecommunication is shown.

10:00–10:15 Break

10:15

2aSPa8. Acoustic video streaming in a waveguide. Philippe Roux, Jit Sarkar, W. A. Kuperman, W. S. Hodgkiss, Hee Chun Song, Tuncay Akal (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0238, jit@mpl.ucsd.edu), and Mark Stevenson (NATO SACLANT Undersea Res. Ctr., La Spezia 19138, Italy)

The use of two vertical arrays (a source and a receiver array) yields the recording of the broadband transfer matrix in a waveguide. Using time-reversal focusing in this geometry, we have experimentally demonstrated at sea the ability to focus on any receiver array element. Furthermore, we have also shown the possibility to achieve simultaneous multiple foci on different elements of the receiver array. Much like the old dot-matrix printers that used linear arrays of pins to create pixels on paper and thus form an image, we now investigate the use of multiple focal spots on the receiver array elements to create pixels in space-time. Vertical resolution is limited by the number of elements in the receiver array, while horizontal resolution is limited by the frequency bandwidth of the transducers. This acoustic TV opens up the possibility of transmitting full-motion video through an acoustic waveguide, which is a potentially robust form of communication as the human eye is able to recognize images even after considerable loss in quality.

10:30

2aSPa9. KauaiEx: Environmental effects on HF acoustic propagation with application to communications. Michael B. Porter, Paul Hursky, Martin Siderius (Ctr. for Ocean Res., Sci. Applications Intl. Corp., 10260 Campus Point Dr., San Diego, CA 92121), Mohsen Badiy (Univ. of Delaware), Jerald Caruthers (Univ. of Southern Mississippi), William S. Hodgkiss, Kaustubha Raghukumar (Scripps Inst. of Oceanogr.), Dan Rouseff, Warren Fox (Univ. of Washington), Christian de Moustier, Brian Calder, Barbara J. Kraft (Univ. of New Hampshire), Keyko McDonald (Space and Naval Warfare Systems Ctr., San Diego, CA), Peter Stein, James K. Lewis, and Subramaniam Rajan (Sci. Solutions, Inc.)

The Kauai Experiment (22 June–9 July 2003) was designed to study high-frequency (8–50 kHz) acoustics in a shallow-water waveguide. In contrast to much of the previous literature, emphasis was placed on multipath arising from multiple boundary interactions. Various participants were interested in different applications; however, a core theme was the role of the environment on acoustic communications. A great deal of effort was made to characterize the environment including the surface wave spectrum, 2D temperature structure along the propagation path, salinity, currents, and bottom properties. Most of these parameters were measured continuously over the 2 weeks of the experiment, providing information on the diurnal cycles. At the same time, extensive acoustic measurements were made using a variety of vertical line arrays, some of which spanned the entire water column. The acoustic measurements included channel probes to characterize the variation of the impulse response. These probes were interleaved with a variety of modulation schemes for communications including noncoherent methods such as MFSK (multifrequency shift keying), and DPSK (differential phase-shift keying), as well as coherent schemes such as QAM (quadrature amplitude modulation), OFDM (orthogonal frequency division modulation), and PPC (passive-phase conjugation) methods. Thus, the experiment provides a vast amount of information relating environment to acoustic propagation to modem performance. This talk will present an overview of key lessons learned to date.

10:45

2aSpa10. Relating ocean dynamics and sea state to time-angle variability of HF waveforms. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716), Stephen E. Forsythe (Naval Underwater Warfare Ctr., Newport, RI 02841-1708), Michael B. Porter (Sci. Application Intl. Corp., San Diego, CA 92121), and The KauaiEx Group^{a)}

One of the objectives of the Kauai experiment was a better understanding of the ocean dynamics effects on the propagation of high-frequency acoustic signals. Due to a unique oceanographic feature of the shallow water region near the Pacific Missile Range Facility in Kauai, a bottom mounted vertical line array containing eight elements was deployed with sufficiently small element spacing to measure the acoustic energy near the bottom. Simultaneous environmental parameters including current, temperature and salinity profiles, directional surface wave spectra, as well as the wind speed and direction above the sea surface were measured. High correlation between the environmental variability and the received acoustic signals is observed. To interpret the results broadband PE and Gaussian beam ray tracing models were utilized. Arrival time-angle statistics are correlated with the environmental variability due to ocean dynamics in this region. It is shown that variations of the sea surface dynamics exhibit different statistical effects than those occurring within the water column. [Work supported by ONR-3210A.] ^{a)}Paul Hursky, Martin Siderius (SAIC), Jerald Caruthers (USM), William S. Hodgkiss, Kaustubha Raghukumar (SIO), Dan Rouseff, Warren Fox (APL-UW), Christian de Moustier, Brian Calder, Barbara J. Kraft (UNH), Keyko McDonald (SPAWARSSC), Peter Stein, James K. Lewis, and Subramaniam Rajan (SSI).

11:00

2aSpa11. Impact of thermocline and seabed variability on underwater acoustic communications. Martin Siderius, Michael Porter (SAIC, Ctr. for Ocean Res., 10260 Campus Pt. Dr., San Diego, CA 92121), Finn Jensen (SACLANT Undersea Res. Ctr., 19138 La Spezia (SP), Italy), and The Kauai Group (SAIC, SPAWAR, UDel, APL-UW, MPL-UCSD, SSI, UNH, USM)

Shallow water acoustic communications experiments were conducted near Kauai in July 2003 and near Capraia and Elba Islands in October 2003. All experiments took place in approximately 100-m water depth but the oceanography and seabed types differed significantly. The Kauai site had a reflective seabed that was combined with highly variable oceanographic conditions that led to performance closely tied to source/receiver geometry. The Capraia site also has a reflective seabed but the winter conditions produced a more mixed water column with a weaker and less variable thermocline than Kauai. The Elba site had nearly the same oceanographic conditions as Capraia but the seabed is a highly lossy. In each of these experiments, signals were transmitted over many hours from fixed and moving platforms and were received at multiple ranges and depths using vertical arrays and single hydrophones. Extensive environmental measurements were made simultaneous to the acoustic transmissions (e.g., measurements of the water column temperature structure and surface wave heights). In this paper, the correlation between environmental factors and communications performance will be presented along with the predictions from modeling. The performance of both multi-frequency shift keying (MFSK) and direct sequence spread spectrum (DSSS) signals will be discussed.

11:15

2aSpa12. Comparison of two underwater acoustic communications techniques for multi-user access. Paul Hursky, T. Martin Siderius, Michael B. Porter, and The KauaiEx Group^{a)} (SAIC, 10260 Campus Point Dr., San Diego, CA 92121)

Frequency hopped frequency shift keying (FHFSK) and code division multiple access (CDMA) are two different modulation techniques for multiple users to communicate with a single receiver simultaneously. In July 2003, these two techniques were tested alongside each other in a shallow

water coastal environment off the coast of Kauai. A variety of instruments were used to measure the prevailing oceanography, enabling detailed modeling of the channel. The channel was acoustically probed using LFM waveforms and m-sequences as well. We will present the results of demodulating the FHFSK and CDMA waveforms and discuss modeling the channel for the purpose of predicting multi-user communications performance. ^{a)}Michael B. Porter, Paul Hursky, Martin Siderius (SAIC), Mohsen Badiey (UD), Jerald Caruthers (USM), William S. Hodgkiss, Kaustubha Raghukumar (SIO), Dan Rouseff, Warren Fox (APL-UW), Christian de Moustier, Brian Calder, Barbara J. Kraft (UNH), Keyko McDonald (SPAWARSSC), Peter Stein, James K. Lewis, and Subramaniam Rajan (SSI).

11:30

2aSpa13. The effect of internal waves on mid-frequency underwater acoustic communications. T. C. Yang (Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375)

Internal waves are abundant in coastal waters and are known to have a significant impact on acoustic signal propagation. The effect includes significant signal fading to loss of signal coherence depending on the acoustic signal frequencies. This paper presents the analysis results using binary phase-shifted keying signals for underwater acoustic communications during the ASCOT01 experiment. We report the measurements of temporal coherence time as a function of time and relate it to the performance of the decision feedback equalizer. The interpacket coherence was measured using the LFM signals in each packet which were separated by ~ 2 min. We see that the intensity and arrival time of the multipath arrivals vary significantly from packet to packets. The signal coherence within a packet (intrapacket) is measured using consecutive *m*-sequences. We find that at mid (2–5 kHz) frequencies, the acoustic environment, while randomly changing with time on the scale of minutes, presents an instantaneous deterministic environment for each packet; the coherence time is much longer than the packet length. The environmental impact can be mitigated by adequate sampling of the channel impulse response function. [Work supported by the ONR.]

11:45

2aSpa14. The effect of internal waves on low-frequency underwater acoustic communications. T. C. Yang (Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375)

Internal waves are abundant in coastal waters and are known to have a significant impact on acoustic signal propagation. The effect includes significant signal fading to loss of signal coherence depending on the acoustic signal frequencies. This paper presents the analysis results using binary phase-shifted keying signals for underwater acoustic communications (ACOMMS) during the SWARM95 experiment. As reported in the open literature, the acoustic data were strongly affected by the presence of internal waves. For example, at 200–400 Hz, the signal characteristics changes drastically at the scale of ~ 2 –6 min. The most noticeable changes include the depth where the signal intensity peaks, the multipath arrival (time) pattern, and the signal amplitude and phase at a fixed depth. At these frequencies, the ACOMMS packet length for a reasonable amount of data will be long (due to the limited bandwidth) and comparable to the signal coherence time. The bit error rate of a packet varies depending on the temporal coherence within a packet (intrapacket). We report the temporal coherence measurements as a function of packet transmission time and relate that to the performance of the channel equalizer. [Work supported by the U.S. ONR.]

2a TUE. AM

Session 2aSPb**Signal Processing in Acoustics and Acoustical Oceanography: Inverse Problems in Seismic Signal Processing**

Max Deffenbaugh, Cochair

Exxon Mobil Upstream Research, P.O. Box 2189, Houston, Texas 77252

Alan W. Meyer, Cochair

*Lawrence Livermore National Laboratories, 700 East Avenue, Livermore, California 94550***Chair's Introduction—9:00*****Invited Papers*****9:05****2aSPb1. Overview of inverse problems in acoustic and seismic signal processing.** Leon H. Sibul and Michael J. Roan (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804-0030, lhs2@psu.edu)

An overview of inverse problems in acoustic and seismic signal processing is presented. The main goal of seismic inversion is to determine earth properties from seismic reflections or signals. Acoustic inversion includes ultrasonic imaging, acoustic tomography, acoustic oceanography, and propagation medium modeling. System identification, parameter estimation, image reconstruction from incomplete data, and blind deconvolution are all inverse problems. Most practical inverse problems are underdetermined and their solutions are not unique. The nonuniqueness is addressed by maximum entropy, minimum relative entropy methods, and Bayesian methods. Bayes method determines maximum *a posteriori* (MAP) probability density estimates of the model. Many inverse problems are numerically unstable. Various regularization methods are used to alleviate numerical instability. Classical least-squares inversion, singular value decomposition, and recent developments of blind deconvolution are reviewed. [Work supported by ONR.]

9:25**2aSPb2. Detection of uncertain seismic signals.** D. B. Harris (Energy and Environment, Earth Sci. Div., Univ. of California, Lawrence Livermore Natl. Lab., 7000 East Ave., Livermore, CA 94550-9234, harris2@llnl.gov)

Current practice in seismic event detection is concentrated at the extremes of a spectrum of possibilities determined by the degree of knowledge available about the signals to be detected. At one end of the spectrum, information is available about only the frequency content of the signal. Correspondingly, simple energy detectors operating in fixed passbands are the detection method of choice: the seismological standard for detection of events over broad regions. At the other end of the spectrum, the signal is completely known. For such cases, correlation detectors (matched filters) are indicated. Correlation detectors are the emerging standard for detection of repetitive events from sources with very limited geographic extent (a few wavelengths), e.g., mines, compact earthquake swarms, and aftershock sequences. It is desirable to seek detectors that operate between the extremes, i.e., that have much of the sensitivity of correlation detectors, but more of the flexibility of energy detectors to detect signals with greater uncertainty. Subspace detectors offer one approach to achieving this trade-off. This talk describes an application of subspace detectors to detect events in a swarm with significant signal variability, and demonstrates the theoretical ability of these detectors to grade almost continuously between correlators and energy detectors.

9:45**2aSPb3. Characterization of seismic signals and background seismic noise.** Robert Uhrhammer (Berkeley Seismological Lab., 215 McCone Hall, Univ. of California, Berkeley, Berkeley, CA 94720-4760)

A variety of signal processing methodologies have been utilized in the analysis and characterization of the seismic signals and the background noise recorded by modern seismic instrumentation at the UC Berkeley Seismological Laboratory. The capabilities of seismic instrumentation have vastly improved over the past couple of decades since the advent of microcomputer and related technologies. Modern seismic instrumentation consists primarily of force feedback broadband seismometers (to record weak ground motions) and strong motion accelerometers (to record large local shocks) coupled to high resolution digital (24-bit+integer) data loggers. We discuss the characteristics and capabilities of the various seismic sensors and the digital recording systems. The earth's background noise and the instrumental self-noise characteristics are discussed. Various weak signals (both coherent and transient) detection and analysis methodologies are also discussed along with some interesting cases of detecting and characterizing unusual natural and artificial seismic sources.

10:20

2aSPb4. Approximate methods for time-reversal processing of large seismic reflection data sets. James Berryman (Lawrence Livermore Natl. Lab, P.O. Box 808, L-200, Livermore, CA 94551-9900)

Seismic reflection data sets, whether collected on land or at sea, contain very large amounts of data even by modern standards. These data are typically sampled regularly in time but irregularly in space. So, temporal Fourier transforms are straightforward, but spatial transforms can be problematic, and very often require special efforts to resample the data onto a uniform grid in space. On the other hand, time-reversal processing of such data requires only the temporal Fourier transform, which is easy for these data. Accounting for data reciprocity, a complex square transfer matrix can be constructed at each frequency, and standard methods can then be used—at least in principle—to find singular values and singular vectors (SVD) of this large matrix. Once those singular vectors having the largest singular values are known, it is relatively straightforward to reconstruct the scattering surfaces giving rise to the data matrix. The biggest leap in this data processing scheme comes from the need to find the SVD for very large complex matrices. But iterative schemes using Krylov space methods can be applied to resolve this difficulty and produce a relatively small number of singular vectors containing the most relevant information.

10:40

2aSPb5. Planarly layered diffraction tomography with accurate Green function. Sean K. Lehman (Lawrence Livermore Natl. Lab., L-154, 7000 East Ave., Livermore, CA 94550)

Diffraction tomography (DT) imaging techniques require knowledge of the background Green function. Due to its simplicity, it is standard practice to use a homogeneous medium Green function. We have developed a model of a planarly layered Green function that can be used in DT imaging of planarly layered media where the layer acoustical properties and dimensions are known. We present the theory and applications. [Work performed under the auspices of the U.S. Department of Energy by University of California, Lawrence Livermore National Laboratory under Contract W-7405-Eng-48.]

11:00

2aSPb6. Estimating lithology and fluid parameters from seismic data. Michael E. Farrell (Exxonmobil Res. Co., P.O. Box 2189, Houston, TX 77252-2189, mike.e.farrell@exxonmobil.com)

A constrained inversion approach is presented for making quantitative estimates of lithology and fluid parameters using compressional wave reflection data (P-to-P) and converted wave reflection data (P-to-S). The method is horizon based, and requires the seismic amplitude information from multiple offsets at a particular zone of interest. Results are discussed from two specific cases. In the first example estimates of lithology are derived from only P-to-P reflection data, assuming the fluid properties are known. This turns out to be a well-posed problem with a unique solution, and is robust in the presence of additive random noise. In the second example both lithology and fluid parameters are estimated using a joint inversion of P-to-P and P-to-S seismic data. This is a much more difficult problem, and does not have a unique solution. However, by including realistic lithology and fluid constraints, good estimates of the parameters can be obtained. Furthermore, adding these constraints to the inversion of only P-to-P wave data also yields a good solution. This suggests that the constraints are more important than doing a joint inversion with the additional P-to-S data.

Contributed Papers

11:20

2aSPb7. Utilization of the interference invariant for geoacoustic parameter inversion in shallow water. Peter M. Daly and Peter N. Mikhalevsky (SAIC Ocean Systems Operation, 1710 SAIC Dr., MS 1-11-15, McLean, VA 22102, peter.m.daly@saic.com)

S. D. Chuprov's interference invariant, or β parameter, characterized broadband striation patterns as a function of range and frequency. Puchenkina and Salin proposed using the β parameter to estimate geoacoustic bottom parameters. Further work by Baggeroer, Rousseff, and Spindel recommended characterizing β as a probability distribution instead of a discrete quantity. The authors of this presentation will illustrate use of the β distribution to perform geoacoustic parameter inversion in shallow water, using simulated annealing methods. This inherently broadband process utilizes ships of opportunity combined with *a priori* range information to provide high-SNR input signals to the inversion process. [Work sponsored by DARPA Advanced Technology Office, Contract N00024-01-C-6319.]

11:35

2aSPb8. Techniques for estimating sediment attenuation from chirp sonar data. Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

Recent broadband (50–500 Hz) matched-field inversions of geoacoustic parameters indicated that sediment attenuation is difficult to estimate especially at short source-receiver distances. On the other hand, geoacoustic inversions become less practical at long source-receiver distances due to the range-dependency of both ocean bottom and water column. Several inversion techniques are investigated to estimate *in-situ* sediment attenuation from normal incidence reflection data, collected by chirp sonars. Both frequency-shift and attenuation roll-off measurement techniques showed nonlinear frequency dependency of attenuation for sandy sediments at the 2-12-kHz frequency band. Estimated attenuation values are extrapolated to lower frequencies by using the Biot theory. The extrapolated results are consistent with those of the matched-field inversions. Several advantages and limitations of the frequency-shift and attenuation roll-off measurement techniques are also discussed. [Work supported by ONR.]

Session 2aUW

Underwater Acoustics: General Underwater Acoustics

Natalia Sidorovskaia, Chair

Department of Physics, University of Louisiana at Lafayette, Lafayette, Louisiana 70504

Chair's Introduction—7:55

Contributed Papers

8:00

2aUW1. New methods in the analysis of resonances. H. Uberall (Dept. of Phys., Catholic Univ. of America, Washington, DC 20064) and M. Werby (NRL Code 7181, Stennis Space Ctr., MS 39529)

From water inclusions to layered elastic shells with included fluids, all may be analyzed by a common method. We demonstrate that the background method is always possible and is because solutions of linear non-homogeneous, differential equations are composed of the supposition of the homogeneous solution plus a particular solution. Moreover, the partial wave solutions that arise in the usual developments have both inertial and damping components. We show that one may use these facts to determine both the resonance locations and the resonance widths and we present a context for which one may give meaning to observed resonance features.

8:15

2aUW2. Acoustical imaging of underwater objects using the bistatic ramp response signals. Wei Li (Natl. Univ. of Singapore, Singapore), Gui-Rong Liu (Natl. Univ. of Singapore, Singapore), and Vijay Varadan (Penn State Univ., University Park, PA 16801, vjvesm@engr.psu.edu)

The T -matrix methods have been successfully employed for underwater acoustical imaging techniques and the inverse analysis of acoustic scattering problems. Most inverse techniques use the backscattering signals in the acoustical far field to retrieve the shape and size information of an underwater object, such as the ramp response technique. This paper addresses a modified ramp response technique, which could be used to reconstruct the 3D image of an object for the bistatic case. This technique shows that the bistatic ramp response is proportional to the profile function of an underwater object based on the small bistatic angle assumption. The numerical examples demonstrate that the bistatic ramp response technique is still valid to obtain an excellent profile function even for the bistatic case with a fairly big bistatic angle. This bistatic ramp response technique allows us to reconstruct the 3D image of an underwater object with only one receiver.

8:30

2aUW3. A theoretical method to extract target information between two vertical arrays. William Sanders and Michael Werby (NRL Code 7181, Stennis Space Ctr., MS 39529)

For a known source and suitable environmental information one may determine the presence (or absence) of a target ensonified by a guided wave initiated by the source by using vertical arrays that sandwich a region of interest. We show first that one can represent the ensonified object as a source term in the solution of an inhomogeneous Sturm–Liouville problem where the homogeneous solution corresponds to the solution in the absence of a target. This allows one to choose a bounded region about the object and evaluate the equivalent of Huygens' integral (over the boundary that includes the vertical arrays), which enables one to trace the signal back to the object. This determines both the range and depth and even the target strength. When this is extended to a pulse signal

then the information leads to higher fidelity than the frequency results due to phase averaging of fluctuations. Thus we demonstrate this to be a good strategy for an experimental study that would lead to a robust detection method that may be implemented on a rapid basis.

8:45

2aUW4. The diffraction of leaky Rayleigh waves at the extremity of a fluid-loaded plate. Nico F. Declercq, A. Teklu, M. A. Breazeale (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, Oxford, MS 38677, NicoF.Declercq@UGent.be), Rudy Briers (KATHO, B-8820 Torhout, Belgium), Oswald Leroy (KU Leuven Campus Kortrijk, B-8500 Kortrijk, Belgium), and Joris Degrieck (Ghent Univ., B-9000 Ghent, Belgium)

This study reveals that leaky Rayleigh waves, when scattered at the extremity of a thick plate swamped in water, generate leaky Rayleigh waves that propagate around the corner. Furthermore, it is experimentally proved that leaky Rayleigh waves are stimulated by the borders of a Gaussian-bounded beam and not by the interior of the beam. A comparison between the scattering of leaky Rayleigh waves and the scattering of Scholte–Stoneley waves at the extremity is also outlined and shows that leaky Rayleigh waves on the vertical edge at the extremity of the plate are best stimulated by means of incident leaky Rayleigh waves and not by means of Scholte–Stoneley waves. [Work supported by The Flemish Institute for the Encouragement of the Scientific and Technological Research in Industry (I.W.T.) and by a NATO Collaborative Linkage Grant.]

9:00

2aUW5. The dependence of Lamb wave stimulation parameters on the impedance difference between upper and lower liquid. Nico F. Declercq, Filip Van den Abele, Joris Degrieck (Soete Lab, Dept. of Mech. Constr. and Prod., Ghent Univ., Sint Pietersnieuwstraat 41, B-9000 Ghent, Belgium, NicoF.Declercq@UGent.be), and Oswald Leroy (IRC-KULAK, B-8500 Kortrijk)

It is known that inhomogeneous plane waves are better capable of stimulating Lamb waves in a solid plate than homogeneous plane waves. For each Lamb mode there exists an angle of incidence and an inhomogeneity parameter that maximizes its stimulation. If for example a solid plate separates two different liquids (a lower liquid and an upper liquid), then the angle of incidence, for stimulating the same Lamb mode, is unchanged, whereas the inhomogeneity differs compared with the situation where both liquids are equal to each other. If for a given upper liquid and for a given Lamb mode this inhomogeneity difference is plotted as a function of the impedance of the lower liquid, then a linear dependence is noticed. It is noticed that all physical liquids follow this linear tendency. Furthermore, this linearity depends on the generated mode and also on the properties of the solid plate. [Work supported by The Flemish Institute for the Encouragement of the Scientific and Technological Research in Industry (I.W.T.)]

9:15

2aUW6. Born approximation for scattering by evanescent waves: Comparison with exact scattering by an infinite fluid cylinder. Philip L. Marston (Phys. Dept., Washington State Univ., Pullman, WA 99164-2814)

In some situations, evanescent waves can be an important component of the acoustic field within the sea bottom. For this reason (as well as to advance the understanding of scattering processes) it can be helpful to examine the modifications to scattering theory resulting from evanescence. Modifications to ray theory were examined in a prior approximation [P. L. Marston, *J. Acoust. Soc. Am.* **113**, 2320 (2003)]. The new research concerns the modifications to the low-frequency Born approximation and confirmation by comparison with the exact two-dimensional scattering by a fluid cylinder. In the case of a circular cylinder having the same density as the surroundings but having a compressibility contrast with the surroundings, the Born approximation with a nonevanescence incident wave gives only monopole scattering. When the cylinder has a density contrast and the same compressibility as the surroundings the regular Born approximation gives only dipole scattering (with the dipole oriented along to the incident wavevector). In both cases when the Born approximation is modified to include the evanescence of the incident wave, an additional dipole scattering term is evident. In each case the new dipole is oriented along to the decay axis of the evanescent wave. [Research supported by ONR.]

9:30

2aUW7. Fluid-filled spheres: Theoretical and measured scattering response. David M. Deveau (Naval Undersea Warfare Ctr., 801 Clematis St., West Palm Beach, FL 33401, deveau@wpb.nuwc.navy.mil)

Investigation into the scattering nature of surfaces or other physical objects often requires the use of measurement systems which cannot always be well controlled. This lack of control can be compensated for by calibrating the resulting measurements against a known target. While these targets can be any object, the goal is to use a target that has a stable scattering response and is independent of angle. The ideal shape is that of a sphere, but even this can be improved with the addition of internal fluids that focus and temperature stabilize the scattering response. A scatter response model of a sphere has been developed and used to design four thin-walled spheres, each with a different diameter and filled with a focusing fill fluid (Fluorolube). One pair of spheres was measured in an ocean environment, while the second pair was tested in a controlled test tank from 5–50 kHz but using shorter continuous wave pulses. While the ocean-measured spheres closely matched the model, the test-tank measurements showed a marked difference from the model. Changes to the model will be explored to determine if theoretical minimums for pulse length are insufficient for targets of this density or focusing capability.

9:45

2aUW8. Scattering from bubbles rising in a vertical line—Comparison between theory and experiments. Aneta Nikolovska, Andrew Ooi (Dept. of Mech. and Manufacturing Eng., The Univ. of Melbourne, Parkville, VIC 3010, Melbourne, Australia), and Richard Manasseh (CSIRO Thermal and Fluids Eng., Highett, VIC 3190, Melbourne, Australia)

In this paper, the experimental data from the sound field around similar sized air bubbles rising in a vertical chain are analyzed. The data reveal a strong anisotropy in the acoustic field. The transition from individual to continuum behavior in a bubble chain is not correctly described with classical theory especially when the bubbles are uniformly sized, discretely populated, and the frequency of interest is close to the natural frequency of the individual bubbles. Single compact scatterers initiated at frequencies near their resonant frequency in isolation act preliminarily as monopole sources, amplifying the local pressure field by a factor of order $1/(k^*a)$, a being the scattering radius and k the wave number at the resonant frequency in the surrounding material [Tolstoy, 1986]. A laboratory investigation used air bubbles in fresh water and varied the bubble sizes and separation by carefully controlling bubble production rates. A linear coupled equation method was developed to explain the result. The model

reproduced the acoustic pressure anisotropy along the chain and the change in pulse waveform along the chain. The results suggest that the enhancement of sound intensity along the chain can be explained by bubbles acting as resonant scatterers retransmitting the acoustic energy.

10:00–10:15 Break

10:15

2aUW9. Development of a submersible, baffled impedance tube for characterization of bubbly liquids. Jed C. Wilbur, Jason Holmes, Michael Canney, William M. Carrey, and Ronald A. Roy (Dept. of Aersp. and Mech. Eng., Boston Univ., Boston, MA 02215)

The ability to measure the acoustic properties of bubbly liquids *in situ* has long been a goal of the underwater acoustics community. An apparatus for characterizing these properties in the 1–16-kHz frequency range capable of submerged deployment in a controlled environment (large tank or test pond) has been developed. A water-filled, 2.5-cm-thick walled, baffled, stainless steel impedance tube was used to measure the complex reflection coefficient at the opening. The classical result for the radiation impedance of a baffled, open tube is verified, as is that for the radiated sound field along the tube axis. The test tank is then filled with air bubbles (order 500 micron mean radius and 0.1% vf) and the measurement repeated at the water/bubbly liquid interface. The acoustic properties of the bubbly liquid were obtained as a function of frequency by an inversion process using classical theory. Because of the high attenuation in bubbly liquids, the finite-sized baffle (61-cm outer diameter) is treated as being infinite in extent. The measured sound speeds and attenuations are compared with established theory. [Work supported by U.S. Navy ONR.]

10:30

2aUW10. Optimal passive source localization in a fluctuating ocean waveguide based on an analytic model for the mean field and covariance. Purnima Ratilal, Ioannis Bertsatos, Tianrun Chen, Michele Zanolin, and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139, purnima@mit.edu)

The ocean acts as an enormously complex channel for signal transmission. It is characterized by temporal fluctuations and spatial variabilities that often lead to significant randomization of the measured field. Signal processing techniques must be derived from fundamental physical models to attain statistical optimality in such an environment. Here we present such an analysis with a new analytic model [Ratilal and Makris, *J. Acoust. Soc. Am.* **114**, 2428 (2003)] for the mean field and its covariance after propagation through a fluctuating ocean waveguide based on the first principles of waveguide scattering theory. The model is advantageous because it includes the primary physical effects of attenuation, dispersion, and coupling of modal energy due to multiple forward scattering in a convenient and intuitive form that is well suited to analytic manipulations. An example will be presented for passive source localization in range and depth in a fluctuating ocean waveguide by nonlinear matched field inversion. The necessary conditions on sample size and signal-to-noise ratio (SNR) for unbiased and minimum variance (optimal) estimates of source position are then derived using asymptotic statistical inference theory [Nafatali and Makris, *J. Acoust. Soc. Am.* **110**, 1917–1930 (2001)]. Degradation in performance by environmental uncertainties are quantified within this statistical framework.

10:45

2aUW11. Some interesting features in pulse propagation for environments with structured velocity profiles. R. Field and M. Werby (NRL Code 7181, Stennis Space Ctr., MS 39529)

On a diurnal time scale the velocity profiles in some shallow water environments can change enough to modify signal detection features. In particular, if a vertical array is used to collect data, then contour plots of

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the detected signals relative to time and vertical distance can reflect features of the environment that are readily interpreted and therefore reveals some details of interest. Two features are investigated here. One pertains to surface ducting, which may be present over some limited time scale on a daily basis. The other feature is a rather pronounced variation in bottom ducting diurnally. At suitable frequencies (1 KH–5 KH) contour plots indicate discrete time arrivals due to sound waves trapped in the surface duct that reflect fast early (precursor) group arrivals. This effect has been studied earlier by H. DeFerrari and can be useful in extracting environmental features. Broad bottom ducts manifest themselves as broad group fronts characterized by almost vertical initial arrivals. This pattern varies on a daily basis. Thus these contours offer a simple method for diurnal duct variations and ought to be of interest for passive detection.

11:00

2aUW12. Recovery and digitization of archival analog ocean acoustic data. Jack Shooter (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029) and Roy D. Gaul (BlueSea Corp.)

In October 2002 the Office of Naval Research (ONR) sponsored a Convocation that reviewed the ocean acoustic studies done from 1967 to 1992 under the U.S. Navy's Long Range Acoustic Propagation Project (LRAPP). Beginning in 1972 LRAPP fielded self-contained assemblies of vertically distributed hydrophones as part of environmental acoustic exercises in a variety of oceanic regions. Analog signals were recorded in a submerged buoy on multi-channel magnetic tape. Presentation of results from a 1975 measurements exercise in the Northeast Pacific stimulated interest in recovering and digitizing the 10-day dataset from 13 hydrophones. ONR sponsored a pilot project to demonstrate the feasibility of recovering the data and setting up digital files while preserving absolute accuracy. The original magnetic tape was sticky. It had to be heat treated and cleaned before playback. A spectrum from the original analysis for near-field passage of a ship was used to validate data recovery in the range of 10–500 Hz. A final objective is to produce calibrated time series so that the digital dataset can be made available for general use. [Work supported by ONR.]

11:15

2aUW13. Passive acoustic localization for Sciaenid habitat in coastal water of Taiwan. Chang Tu, Ruey-Chang Wei (Inst. of Undersea Technol., Natl. Sun Yat-sen Univ., 70, Lien-hai Rd., Kaohsiung, Taiwan, rcwei@mail.nsysu.edu.tw), and Hsiang-Chih Chan (Natl. Taiwan Univ., Taipei 804, Taiwan)

There are many Sciaenid species found in coastal water of Taiwan, and most of them can generate sound in the spawning season. However, due to overfishing, the populations of these high economic value fishes have been greatly decreased. To study and protect Sciaenid, whose habitat should be identified, seasonal protection zones should be set up. In this study, easy and low-cost measure of using passive sonar is proposed to map the habitat in the field. Measurement and analysis of Sciaenid sound were performed to investigate its acoustic characteristic, and frequency range was found to be 350 to 1500 Hz. The statistical distribution of sound levels can be built up by enough field measurement; therefore source levels can be defined in this method. In addition, the transmission loss of in-site shallow water was studied to compute the real decaying factor ($TL = n \cdot \log R$) of the survey site. By using three sonobuoys, with known source level distribution and transmission loss, the distance of possible habitat can then be inverted. Through enough measurements, the map of coastal habitat of Sciaenid can be generated with efficiency and accuracy.

11:30

2aUW14. Development of net cage acoustic alarm system. Shih-Wei Hong, and Ruey-Chang Wei (Inst. of Undersea Technol., Natl. Sun Yat-sen Univ., 70, Lien-hai Rd., Kaohsiung, Taiwan, rcwei@mail.nsysu.edu.tw)

In recent years, the fishery production has been drastically decreased in Taiwan, mainly due to overfishing and coast pollution; therefore, fishermen and corporations are encouraged by government to invest in ocean net cage aquaculture. However, the high-price fishes in the net cage are often coveted, so incidences of fish stealing and net cage breaking were found occasionally, which cause great economical loss. Security guards or a visual monitoring system has limited effect, especially in the night when these intrusions occur. This study is based on acoustic measure to build a net cage alarm system, which includes the sonobuoy and monitor station on land. The sonobuoy is a passive sonar that collects the sounds near the net cage and transmits the suspected signal to the monitor station. The signals are analyzed by the control program on the personal computer in the monitor station, and the alarms at different stages could be activated by the sound levels and durations of the analyzed data. To insure long hours of surveillance, a solar panel is applied to charge the battery, and a photodetector is used to activate the system.

TUESDAY MORNING, 25 MAY 2004

CONFERENCE ROOM L, 8:00 TO 8:50 A.M.

Meeting of the Standards Committee Plenary Group

to be held jointly with the

ANSI-Accredited U.S. Technical Advisory Group (TAG) Meetings for: ISO/TC 43 Acoustics ISO/TC 43/SC 1 Noise, and IEC/TC 29 Electroacoustics

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S2, S3, and S12, which are scheduled to take place later on the same day. Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all four S Committees.

The ANSI-Accredited U.S. Technical Advisory Group (TAGs) for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, whose membership consists of members of S1 and S3, and other persons not necessarily members of these Committees, will meet during the Standards Plenary meeting. The ANSI-Accredited US Technical Advisory Group (TAG) for ISO/TC 43/SC 1 Noise, whose membership consists of the members of S12 and other persons not necessarily members of S12, will meet as well. *The reports of the Chairs*

of these TAGs will not be presented at any other S Committee meeting. There will be a report on the interface of S1 and S3 activities with those of ISO/TC 43 and IEC/TC 29 including plans for future meetings of ISO/TC 43 and IEC/TC 29.

Members of S2 Mechanical Vibration and Shock (and U.S. TAG for ISO/TC 108 and four of its Subcommittees, (SC2, SC3, SC5, and SC6) are also encouraged to attend the Standards Committee Plenary Group meeting even though the S2 meeting will take place later in the day.

TUESDAY MORNING, 25 MAY 2004

CONFERENCE ROOM L, 9:00 TO 10:00 A.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 8:00 a.m. on Tuesday, 25 May 2004.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

TUESDAY MORNING, 25 MAY 2004

CONFERENCE ROOM L, 10:30 TO 12:00 NOON

Meeting of Accredited Standards Committee (ASC) S12 Noise

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort and physical acoustics as related to environmental and occupational noise.

TUESDAY AFTERNOON, 25 MAY 2004

ROYAL BALLROOM A, 2:00 TO 3:25 P.M.

Session 2pAAa

Architectural Acoustics: Methods to Quantify Opera House Acoustics III: Simulations and Auralizations

Jin Yong Jeon, Chair

School of Architectural Engineering, Hanyang University, 12 Haengding-dong, Seongdong-gu, Seoul 133-79, Korea

Chair's Introduction—2:00

Invited Papers

2:05

2pAAa1. An experimental comparative study of 20 Italian opera houses: Measurement techniques. Angelo Farina, Enrico Armelloni, and Paolo Martignon (Indust. Eng. Dept., Univ. of Parma, Via delle Scienze 181/A, 43100 Parma, Italy)

For “acoustical photography” we mean a set of measured impulse responses, which enable us to “listen” at the measured room by means of advanced auralization methods. Once these data sets have been measured, they can be employed in two different ways: objective analysis and listening test. In fact, it is possible to compute dozens of acoustical objective parameters, describing the temporal texture, the spatial effect and the frequency-domain coloring of each opera house. On the other hand, by means of the

auralization technique, it becomes easy to conduct listening experiments with human subjects. This paper focuses principally on the development and specification of the measurement technique, which is the topic assigned to the research unit of Parma, to which the authors belong. It describes the hardware equipment, the software, the electro-acoustic transducers (microphones and loudspeakers), the measurement positions, the system for automatic displacement of the microphones and the conditions of the room during the measurements. Experimental results are reported about a couple of opera houses which were employed for testing the measurement procedure and showing the benefits of the new method against the previously employed ones.

2:25

2pAAa2. Auditorium acoustics evaluation based on simulated impulse response. Shuoxian Wu, Hongwei Wang, and Yuezhe Zhao (Dept. of Architecture, South China Univ. of Technol., Guangzhou 510640, China)

The impulse responses and other acoustical parameters of Huangpu Teenager Palace in Guangzhou were measured. Meanwhile, the acoustical simulation and auralization based on software ODEON were also made. The comparison between the parameters based on computer simulation and measuring is given. This case study shows that auralization technique based on computer simulation can be used for predicting the acoustical quality of a hall at its design stage.

2:45

2pAAa3. Calculating auditorium acoustic parameters from measured binaural impulse response. Yuezhe Zhao and Shuoxian Wu (Dept. of Architecture, South China Univ. of Technol., Guangzhou 510640, China)

The differences between Chinese traditional music and western music will cause some specific acoustics requirements for the auditoriums where national music programs are played. There are varieties of coupled spaces in buildings. These coupled spaces have some special acoustical characteristics. As a starting point of these researches, the binaural impulse responses and other acoustical parameters were measured and calculated for Guangzhou Southern Theatre where Chinese traditional music and Guangdong folk music are usually performed. The measurement results were analyzed in detail herein.

3:05

2pAAa4. Acoustical measurements in a 1:10 scale model of a multi-purpose hall. Jin Yong Jeon and Jong Kwan Ryu (School of Architectural Eng., Hanyang Univ., Seoul 133-791, Korea)

Measurements were made in a 1:10 scale model of a 1500-seat multi-purpose hall to investigate the acoustic properties of its opera mode. The length from the stage front to the rear wall is 33 m and its maximum width is 28 m with a volume of 14300 m³. Reflecting surfaces were included to scatter early energy from the stage area onto the audience areas. Several settings of the sources were used on the platform and in the pit in order to determine how a lack of brilliance caused by diffraction occurred in the stalls audience. The advantages of the curved ceiling surface in front of the proscenium above the orchestra pit were exploited. In addition, the effect of some diffusing treatments on surfaces adjacent to the pit and on the side walls was also investigated.

TUESDAY AFTERNOON, 25 MAY 2004

ROYAL BALLROOM A, 3:30 TO 5:15 P.M.

Session 2pAAb

Architectural Acoustics: Methods to Quantify Opera House Acoustics IV: Measurements

Richard H. Campbell, Chair

Bang Campbell Associates, 3 Water Street, Woods Hole, Massachusetts 02543

Chair's Introduction—3:30

Invited Papers

3:35

2pAAb1. Methods to quantify opera houses acoustics—Changes to why, what, and how? Robert W Harris (Arup Acoust., Parkin House, 8 St. Thomas St., Winchester SO23 9HE, UK, rob.harris@arup.com)

The historical and current reasons for quantifying opera house acoustics (the why?) are reviewed. The emerging shift from quantitative assessment (by measurement) to qualitative assessment (by auralization) is discussed. The key characteristics of the acoustics of opera houses (the what?) are then considered. Assessment of a theatre with an audience present is suggested to be more important than unoccupied measurements. This has implications for the development of assessment methods. The suitability of classical room acoustics parameters—mostly derived from concert hall studies—is debated (the how?). Possible new assessment techniques are emerging. Characteristics not addressed by the classical parameters include the early (ensemble) and late (room) support provided to the singers and the onset threshold of image shifting of the sound of instruments within the orchestra pit. The assessment of loudness balance between the stage and the pit requires refinement and development. A difficulty with the assessment and comparison of opera houses is the effect of the stage settings on the acoustics.

2pAAb2. Design of a new stage shell for the Stadthaus in Winterthur, Switzerland. Eckhard Kahle (Kahle Acoust., 156 Ave. du Parc, B-1190 Brussels, Belgium, kahle@skynet.be) and Brian F. G. Katz (LIMSI-CNRS, F-91403 Orsay, France)

A new stage shell was installed in the Stadthaus, a 700-seat concert hall in Winterthur which was originally designed by architect Gottfried Semper. Musicians of the local symphony orchestra had been complaining about poor on-stage listening conditions in the stage house that was later added to the room. The small stage house (width 12 m, depth 7 m, height 7 m) was overly loud and reverberant. The design of the new stage shell will be presented and the acoustic results, both objective and subjective, will be presented. Comments by the musicians and audience members show that a stage shell must provide more than just efficient projection into the audience area. Absorption placed at the rear of the stage is paramount for obtaining proper orchestral balance, good on-stage hearing condition and clarity of sound.

Contributed Papers

4:15

2pAAb3. Measurements of the self-to-other ratio in an opera chorus in performance. Sten Ternstrom (Dept. of Speech, Music & Hearing, KTH, Stockholm, sten@speech.kth.se) and Pamela Davis (Univ. of Sydney, Sydney, Australia)

The objective of this study was to obtain measurements of the self-to-other ratio (SOR) in the opera chorus on stage. Four members of the opera chorus of Opera Australia volunteered to carry wireless binaural microphones during a dress rehearsal of Verdi's *The Masked Ball*, in the Sydney Opera House. Conditions were those of actual performance in costume, with soloists and orchestra. Hence, the recordings are very realistic, if acoustically not so stringently controlled. The SOR was estimated from the recordings using M/S matrixing, complemented with L-R correlation by adaptive filtering. The results indicate how much louder the singers heard themselves relative to the rest of the ensemble, including the orchestra. The SOR varied with score and on-stage formation, but was 4–8 dB higher than in ordinary choirs (not opera), as might be expected from the podium acoustics. While each opera chorus artist can hear his or her own voice very well, they often hear less of the rest of the chorus and very little of the orchestra. This was borne out in informal listening; the orchestra often became inaudible once the choir entered its louder passages. [Work supported by the Australian Research Council under the SPIRT program.]

4:30

2pAAb4. The measurements technique of balance in opera houses: A study of the sources involved. Linda Parati, Roberto Pompoli, and Nicola Prodi (Dept. of Eng., Univ. of Ferrara, 44100 Ferrara, Italy, lparati@ing.unife.it)

The study of the acoustical balance between the singer and the orchestra by means of room acoustical measurements has shown that the directional characteristics of the source on the stage are important. This investigation compares the performance of two directional and one omnidirectional loudspeakers in emulating a soprano voice. Directivity measurements of a soprano singer were carried out in an anechoic chamber and used as a basis for comparison in room acoustic simulations of the Royal Theatre of Copenhagen, the Ankara Congress and Cultural Center, and the Alberta Jubilee Auditorium. In particular, the balance measurement was simulated and the performance of the different sources with respect to the soprano was assessed.

4:45

2pAAb5. Acoustical measurements in ancient Roman theatres. Andrea Farnetani, Patrizio Fausti, Roberto Pompoli, and Nicola Prodi (Eng. Dept., Univ. of Ferrara, Via Saragat 1, 44100 Ferrara, Italy, afarnetani@ing.unife.it)

The Greek and Roman theatres are among the most precious and spectacular items of cultural heritage in the Mediterranean countries. The theatres are famous not only for their impressive architecture, but also for the acoustic qualities. For this reason it is important to consider these theatres as an acoustical heritage and to study their sound field. Within the activities of the ERATO (identification Evaluation and Revival of the Acoustical heritage of ancient Theatres and Odea) project, acoustical measurements were taken in well-preserved ancient Roman theatres at Aspendos (Turkey) and Jerash (Jordan). Roman theatres have an impressive stage building that forms a back wall in the orchestra area, and it was found that, from the analysis of the acoustical parameters, the reverberation time (e.g., 1.7 s at middle frequencies in the theatre of Aspendos) is quite long compared not only with other open-space theatres but also with closed spaces. Contrary to modern halls the clarity is high and this fact, together with a low sound level in most of the seats, gives the sound field a unique character.

5:00

2pAAb6. Acoustical measurement of the Salt Lake Mormon Tabernacle. Sarah Rollins and Timothy W. Leishman (N283 Eyring Sci. Ctr., Brigham Young Univ., Provo, UT 84602)

An acoustical survey of the Salt Lake Mormon Tabernacle has been performed to assess the behavior of the hall in its current state. The tabernacle is a well-known historical building with a large elongated dome ceiling. This paper discusses the measurements used to characterize the hall. Several parameters derived from omnidirectional, directional, and binaural impulse response measurements are presented. Color maps of the parameters over audience seating areas reveal their spatial variations. These maps and the statistical properties of the parameters aid in clarifying the acoustical characteristics and anomalies of the hall.

Session 2pAAc**Architectural Acoustics: Theaters for Drama Performance: Another Two Decades (1984–2004)
(Poster Session)**

Gregory Miller, Chair

*The Talaske Group, 105 N. Oak Park Drive, Oak Park, Illinois 60301****Contributed Papers***

All posters will be on display from 1:30 p.m. to 5:30 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:30 p.m.

2pAAc1. Denver Center for the Performing Arts, Bonfils Theatre Complex. R. Lawrence Kirkegaard (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc7. University of Cincinnati, College Conservatory of Music. Joseph Kirkegaard (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc2. Portland Performing Arts Center. R. Lawrence Kirkegaard (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc8. Berkeley Repertory RODA Theater. Jason R. Duty and David R. Schwind (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, acoustics@cmsalter.com)

2pAAc3. Broward Center for the Performing Arts. R. Lawrence Kirkegaard (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc9. Aronoff Center. Dawn Schuette (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc4. Guthrie Theatre. Joseph Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc10. University of Alabama at Birmingham, Alys Robinson Stephens Performing Arts Center. Joseph Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc5. Oregon Shakespeare Festival. Joseph Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc11. North Shore Center for the Performing Arts. Dawn Schuette (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc6. Blumenthal Center for the Performing Arts. R. Lawrence Kirkegaard (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc12. Milton Keynes Theatre, Milton Keynes, UK. Rafal J. Orłowski and Rob Harris (Arup Acoust., St. Giles Hall, Pound Hill, Cambridge CB3 0AE, UK)

2pAAc13. Babson College Campus Center. Joseph Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc14. Barbican Centre, Royal Shakespeare Company Theatre. Carl Giegold (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc15. Overture Center. Joseph Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, acoustics@kirkegaard.com)

2pAAc16. Hampstead Theatre, London. Ian Thompson (Arup Acoust., St. Giles Hall, Pound Hill, Cambridge CB3 0AE, UK)

2pAAc17. Founders Theater. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc18. South Dakota State University Theater. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc19. Courtyard Theatre. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc20. Lucille Little Theater Transylvania University. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc21. Ohio Theater. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc22. Williams College Theatre. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc23. Regent Theatre. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc24. Southern Theatre. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc25. Stanley Theatre. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc26. University School Theater. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc27. Yamaguchi Center for Arts and Media, Studio A. Makoto Ino and Keiji Oguchi (Nagata Acoustics, Inc., Hongo-Segawa Bldg., 3F, 2-35-10 Hongo Bunkyo-ku, Tokyo 113-0033, Japan)

2pAAc28. Playwright's Horizon. L. Gerald Marshall (Marshall KMK, 59 S. Greeley Ave., Chappaqua, NY 10514) and K. Anthony Hoover (Cavanaugh Tocci Assoc., Inc., Sudbury, MA 01776)

2pAAc29. Four Rivers Center for the Performing Arts. Katharine Sawicki (Artec Consultants, Inc., 114 W. 26th St., New York, NY 10001, ks@artec-usa.com)

2pAAc30. Toyota City Noh Theater. Keiji Oguchi and Katsuji Naniwa (Nagata Acoustics, Inc., Hongo-Segawa Bldg., 3F, 2-35-10 Hongo Bunkyo-ku, Tokyo 113-0033, Japan)

2pAAc31. Curtis M. Phillips Center for the Performing Arts, University of Florida, Gainesville. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

2pAAc32. Kitakyusyu Performing Arts Center, Theatre. Keiji Oguchi and Katsuji Naniwa (Nagata Acoustics, Inc., Hongo-Segawa Bldg., 3F, 2-35-10 Hongo Bunkyo-ku, Tokyo 113-0033, Japan)

2pAAc33. Bromsgrove Arts Center. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc34. Ruth Shapiro Theater, Carl and Ruth Shapiro Student Center, Brandeis University. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

2pAAc35. Theater 315—Salvation Army Times Square. Ted Pyper (Artec Consultants, Inc., 114 W. 26th St., New York, NY 10001, tap@artec-usa.com)

2pAAc36. Skirball Center for the Performing Arts. Ted Pyper (Artec Consultants, Inc., 114 W. 26th St., New York, NY 10001, tap@artec-usa.com)

2pAAc37. The Chan Centre for the Performing Arts—Telus Studio Theatre. Katharine Sawicki (Artec Consultants, Inc., 114 W. 26th St., New York, NY 10001, ks@artec-usa.com)

2pAAc38. Roundhouse—Royal Shakespeare Company. Alicia Koledin (Acoustic Dimensions, 145 Huguenot St., Ste. 406, New Rochelle, NY 10801, info@acousticdimensions.com)

2pAAc39. Art Sphere Theater. Yasuhisa Toyota and Katsuji Naniwa (Nagata Acoustics, Inc., U.S. Office, 1610 Broadway, Ste. A, Santa Monica, CA 90404)

2pAAc40. Interlochen Center for the Arts—Harvey Theatre. Todd L. Brooks (Artec Consultants, Inc., 114 W. 26th St., New York, NY 10001, tlb@artec-usa.com)

2pAAc41. Fox Cities Performing Arts Center. Todd L. Brooks (Artec Consultants, Inc., 114 W. 26th St., New York, NY 10001, tlb@artec-usa.com)

2pAAc42. Performing Arts Center, The Groton School. Benjamin Markham (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138)

2pAAc43. Tokyo Metropolitan Art Space's Theater. Satoru Ikeda, Keiji Oguchi, Makoto Ino, and Minoru Nagata (Nagata Acoustics, Inc., Hongo-Segawa Bldg., 3F, 2-35-10 Hongo Bunkyo-ku, Tokyo 113-0033, Japan)

2pAAc44. Shizuoka Arts Theatre. Satoru Ikeda, Chiaki Ishiwata, and Motoo Komoda (Nagata Acoustics, Inc., Hongo-Segawa Bldg., 3F, 2-35-10 Hongo Bunkyo-ku, Tokyo 113-0033, Japan)

2pAAc45. Ishikawa Ongakudo's Hougaku Hall. Satoru Ikeda, Keiji Oguchi, and Masaya Uchida (Nagata Acoustics, Inc., Hongo-Segawa Bldg., 3F, 2-35-10 Hongo Bunkyo-ku, Tokyo 113-0033, Japan)

2pAAc46. Princess of Wales Theatre. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada, jokeefe@aercoustics.com)

2pAAc47. Evergreen Cultural Centre. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc48. Ruth Seaton James Centre for the Performing Arts. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc49. A. E. Rawlinson Centre for Arts. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc50. Hummingbird Centre for the Performing Arts. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc51. Theatre Aquarius. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc52. Victoria Hall. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc53. Teatro Sesc Santo Amaro, Sao Paulo, Brazil. Jose Augusto Nepomuceno (Acustica & Sonica, Rua Fradique Coutinho, 955 sala 01 05416-011, Sao Paulo SP, Brazil, janepomuceno@yahoo.com)

2pAAc54. Teatro de Araras, Araras, Brazil. Jose Augusto Nepomuceno (Acustica & Sonica, Rua Fradique Coutinho, 955 sala 01 05416-011 Sao Paulo SP, Brazil, janepomuceno@yahoo.com)

2pAAc55. Teatro Sesc Santana, Sao Paulo, Brazil. Jose Augusto Nepomuceno (Acustica & Sonica, Rua Fradique Coutinho, 955 sala 01 05416-011, Sao Paulo SP, Brazil, janepomuceno@yahoo.com)

2pAAc56. Teatro Sao Pedro, Sao Paulo, Brazil. Jose Augusto Nepomuceno (Acustica & Sonica, Rua Fradique Coutinho, 955 sala 01 05416-011, Sao Paulo SP, Brazil, janepomuceno@yahoo.com)

2pAAc57. Showplace Performance Centre. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc58. The Esplanade. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc59. Teatros dos CEUs, Sao Paulo, Brazil. Jose Augusto Nepomuceno (Acustica & Sonica, Rua Fradique Coutinho, 955 sala 01 05416-011, Sao Paulo SP, Brazil, janepomuceno@yahoo.com)

2pAAc60. Young Centre for the Performing Arts. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc61. Teatro Alfa, Sao Paulo, Brazil. Jose Augusto Nepomuceno (Acustica & Sonica, Rua Fradique Coutinho, 955 sala 01 05416-011, Sao Paulo SP, Brazil, janepomuceno@yahoo.com)

2pAAc62. Chinese Cultural Centre in Toronto. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc63. Queen Elizabeth Theatre. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2pAAc64. New Amsterdam Theater. Paul Scarbrough (Akustiks, 11 N. Main St., Norwalk, CT 06854, pscarbrough@akustiks.net) and David Greenberg (Creative Acoustics, Westport, CT 06880-6134^{a)})

^{a)}Paul Scarbrough was principal-in-charge and David Greenberg was the project manager for Jaffe Holden Scarbrough Acoustics.

2pAAc65. New Victory Theater. Paul Scarbrough and Anthony Nittoli (Akustiks, 11 N. Main St., Norwalk, CT 06854, pscarbrough@akustiks.net^{a)})

^{a)}Paul Scarbrough was the principal-in-charge for Jaffe Holden Scarbrough Acoustics. Anthony Nittoli was senior project manager for Pro-Mix.

2pAAc66. Isabel Bader Theatre. John O'Keefe (50 Ronson Dr., Ste. 165, Toronto, ON, Canada)

2p TUE. PM

Session 2pAAd**Architectural Acoustics and Engineering Acoustics: Bell Labs and Acoustics**

Gary W. Elko, Cochair

Avaya Labs, Basking Ridge, New Jersey 07920

Neil A. Shaw, Cochair

Menlo Scientific Acoustics, Inc., P.O. Box 1610, Topanga, California 90290-1610

James E. West, Cochair

*Department of Electrical and Computer Engineering, Johns Hopkins University, 3400 North Charles Street, Baltimore, Maryland 21218-2686***Chair's Introduction—1:00*****Invited Papers*****1:05****2pAAd1. Communication acoustics in Bell Labs.** J. L. Flanagan (Rutgers Univ., 96 Frelinghuysen Rd., Piscataway, NJ 08854-8088)

Communication acoustics has been a central theme in Bell Labs research since its inception. Telecommunication serves human information exchange. And, humans favor spoken language as a principal mode. The atmospheric medium typically provides the link between articulation and hearing. Creation, control and detection of sound, and the human's facility for generation and perception are basic ingredients of telecommunication. Electronics technology of the 1920s ushered in great advances in communication at a distance, a strong economical impetus being to overcome bandwidth limitations of wireline and cable. Early research established criteria for speech transmission with high quality and intelligibility. These insights supported exploration of means for efficient transmission—obtaining the greatest amount of speech information over a given bandwidth. Transoceanic communication was initiated by undersea cables for telegraphy. But these long cables exhibited very limited bandwidth (order of few hundred Hz). The challenge of sending voice across the oceans spawned perhaps the best known speech compression technique of history—the Vocoder, which parametrized the signal for transmission in about 300 Hz bandwidth, one-tenth that required for the typical waveform channel. Quality and intelligibility were grave issues (and they still are). At the same time parametric representation offered possibilities for encryption and privacy inside a traditional voice bandwidth. Confidential conversations between Roosevelt and Churchill during World War II were carried over high-frequency radio by an encrypted vocoder system known as Sigsaly. Major engineering advances in the late 1940s and early 1950s moved telecommunications into a new regime—digital technology. These key advances were at least three: (i) new understanding of time-discrete (sampled) representation of signals, (ii) digital computation (especially binary based), and (iii) evolving capabilities in microelectronics that ultimately provided circuits of enormous complexity with low cost and power. Digital transmission (as exemplified in pulse code modulation—PCM, and its many derivatives) became a telecommunication mainstay, along with switches to control and route information in digital form. Concomitantly, storage means for digital information advanced, providing another impetus for speech compression. More and more, humans saw the need to exchange speech information with machines, as well as with other humans. Human-machine speech communication came to full stride in the early 1990s, and now has expanded to multimodal domains that begin to support enhanced naturalness, using contemporaneous sight, sound and touch signaling. Packet transmission is supplanting circuit switching, and voice and video are commonly being carried by Internet protocol.

1:25**2pAAd2. Digital signal processing at Bell Labs—Foundations for speech and acoustics research.** Lawrence R. Rabiner (CAIP Ctr., Rutgers Univ., 96 Frelinghuysen Rd., Piscataway, NJ 08854-8088, lrr@caip.rutgers.edu)

Digital signal processing (DSP) is a fundamental tool for much of the research that has been carried out of Bell Labs in the areas of speech and acoustics research. The fundamental bases for DSP include the sampling theorem of Nyquist, the method for digitization of analog signals by Shannon *et al.*, methods of spectral analysis by Tukey, the cepstrum by Bogert *et al.*, and the FFT by Tukey (and Cooley of IBM). Essentially all of these early foundations of DSP came out of the Bell Labs Research Lab in the 1930s, 1940s, 1950s, and 1960s. This fundamental research was motivated by fundamental applications (mainly in the areas of speech, sonar, and acoustics) that led to novel design methods for digital filters (Kaiser, Golden, Rabiner, Schafer), spectrum analysis methods (Rabiner, Schafer, Allen, Crochiere), fast convolution methods based on the FFT (Helms, Bergland), and advanced digital systems used to implement telephony channel banks (Jackson, McDonald, Freeny, Tewksbury). This talk summarizes the key contributions to DSP made at Bell Labs, and illustrates how DSP was utilized in the areas of speech and acoustics research. It also shows the vast, worldwide impact of this DSP research on modern consumer electronics.

1:45

2pAAd3. Speech coding research at Bell Laboratories. Bishnu S. Atal (Dept. of Elec. Eng., Univ. of Washington, Seattle, WA 98195 and Atal-Res., Mukilteo, WA 98275)

The field of speech coding is now over 70 years old. It started from the desire to transmit voice signals over telegraph cables. The availability of digital computers in the mid 1960s made it possible to test complex speech coding algorithms rapidly. The introduction of linear predictive coding (LPC) started a new era in speech coding. The fundamental philosophy of speech coding went through a major shift, resulting in a new generation of low bit rate speech coders, such as multi-pulse and code-excited LPC. The semiconductor revolution produced faster and faster DSP chips and made linear predictive coding practical. Code-excited LPC has become the method of choice for low bit rate speech coding applications and is used in most voice transmission standards for cell phones. Digital speech communication is rapidly evolving from circuit-switched to packet-switched networks to provide integrated transmission of voice, data, and video signals. The new communication environment is also moving the focus of speech coding research from compression to low cost, reliable, and secure transmission of voice signals on digital networks, and provides the motivation for creating a new class of speech coders suitable for future applications.

2:05

2pAAd4. Hands-free human-machine interaction with voice. B. H. Juang (Georgia Inst. of Technol., Atlanta, GA 30332)

Voice is natural communication interface between a human and a machine. The machine, when placed in today's communication networks, may be configured to provide automation to save substantial operating cost, as demonstrated in AT&T's VRCP (Voice Recognition Call Processing), or to facilitate intelligent services, such as virtual personal assistants, to enhance individual productivity. These intelligent services often need to be accessible anytime, anywhere (e.g., in cars when the user is in a hands-busy-eyes-busy situation or during meetings where constantly talking to a microphone is either undesirable or impossible), and thus call for advanced signal processing and automatic speech recognition techniques which support what we call "hands-free" human-machine communication. These techniques entail a broad spectrum of technical ideas, ranging from use of directional microphones and acoustic echo cancellation to robust speech recognition. In this talk, we highlight a number of key techniques that were developed for hands-free human-machine communication in the mid-1990s after Bell Labs became a unit of Lucent Technologies. A video clip will be played to demonstrate the accomplishment.

2:25

2pAAd5. On my association with Bell Labs. M. Mohan Sondhi (Avaya Labs, 233 Mount Airy Rd., Basking Ridge, NJ 07920)

I joined the Acoustics Research department at Bell Labs in 1962, just eight days before AT&T launched the first communications satellite, Telstar. During the 39 years between 1962 and my retirement in 2001, I worked on several problems related in one way or another to the processing of speech signals. Schroeder and Flanagan are presenting talks from a broad perspective in this session, so I will confine this talk to just my own contributions and collaborations for some of the topics on which I worked, e.g., echo cancellation, inverse problems in acoustics, speech analysis, synthesis, and recognition. I will tell you about one of these contributions that fortunately turned out to yield considerable profits to AT&T. To give you a flavor of the spirit of free inquiry at Bell Labs during that period, I will tell you about the contribution that I am most proud of (which was supported for several years even though it had no monetary value). And I will also mention the contribution that is most often cited of all my papers (which was in collaboration with two mathematicians, and had nothing at all to do with acoustics).

2:45

2pAAd6. Speech and hearing acoustics at Bell Labs. Jont Allen (Univ. of Illinois, Urbana, IL, jba@auditorymodels.org)

A. G. Bell's interest in basic research of speech and hearing was one of the keys to the Bell Lab culture. When the first network circuits were built, speech quality was very low. Research was needed on speech articulation (the probability correct for nonsense speech sounds). George Campbell, a mathematician and ultimate engineer, and expert on Heaviside, extended work of Lord Rayleigh. In 1910 Campbell was the first to generate consonant identification confusion matrices, and show sound grouping (features). Crandall took up this work and attempted (but failed) to define the articulation density over frequency. By 1921 Fletcher had solved Crandall's problem, with the the Articulation Index theory, based on the idea of independent feature perception, across frequency and time. In 1929 he wrote his first book, *Speech and Hearing*, which sold over 5000 copies. His second book, *Speech and Hearing in Communications*, was first released in 1953, after his retirement. Other key people that worked closely with Fletcher were J. C. Steinberg, Munson, French, Galt, Hartley, Kingsbury, Nyquist, Sivian, White, and Wegel. I will try to introduce each of these people and describe their contributions to the speech and hearing field.

3:05–3:20 Break

3:20

2pAAd7. Concert hall acoustics. Manfred Schroeder (Drittes Physikalisches Institut, Univ. Goettingen, Goettingen, Germany)

I will review some work at Bell Laboratories on artificial reverberation and concert hall acoustics including Philharmonic Hall (Lincoln Center for the Performing Arts, New York). I will also touch on sound diffusion by number-theoretic surfaces and the measurement of reverberation time using the music as played in the hall as a "test" signal.

2p TUE. PM

3:40

2pAAd8. Transducer research at Bell Labs until 1975. Gerhard M. Sessler (Univ. of Technol., Merckstrasse 25, 64283 Darmstadt, Germany)

Transducer work in the Acoustics Research Department of Bell Labs from about 1960 to 1975 was focussed on electret-based devices. The first such systems were electret headphones which demonstrated the usefulness of this principle in the sense that no dc-bias was required and that the systems were very sensitive, of high quality, and of simple design. This was followed by the invention of the foil-electret microphone in 1962. Initially, these transducers had polyester electrets which, after a thermal charging procedure, kept their charge only for periods of the order of several months. A search for more stable electrets resulted in 1964 in the discovery of fluoroethylenepropylene as a superior material with time constants of the charge decay of hundreds of years under normal environmental conditions. This electret material is still in use in today's transducers. Subsequent activities were directed toward a better understanding of the solid-state properties of these materials. This included, among others, studies of thermally stimulated processes, irradiation effects, and new charging techniques. Later work on transducers centered on the design and characterization of directional microphones, microphone arrays, ultrasonic transducers, touch pads, and other devices useful in telecommunications.

4:00

2pAAd9. Transducer research at Bell Labs 1975 and beyond. James E. West and Ilene Busch-Vishniac (Johns Hopkins Univ., Baltimore, MD, jimwest@jhu.edu)

Condenser microphones, invented at Bell Labs by E. C. Wente and reported in the literature in 1917, are well known for their flat frequency and phase response over a broad range, which make them useful whenever high-quality sound measurements or recordings are required. Two drawbacks, the required high-dc bias and the cost of construction, prevent them from being applied in commercial products such as communication devices, toys, hearing aids, and sound-recording devices. The electret condenser microphone (ECM) eliminates both of the above drawbacks by replacing the needed dc bias with a thin polymer film that is given a permanent charge, the electrical analog of a permanent magnet. Cost of construction is reduced mainly because there is no need to guard against high-voltage breakdown. This technology enables the construction of one- and two-dimensional arrays and up to third-order differential systems for both near- and far-field applications. ECMs are also pliable to most any shape or size, so the construction of continuous strip arrays, transducers for cardiovascular monitoring, MEMS devices, and touch sensors becomes practical. We will discuss some of the experimental systems developed at Bell Labs.

4:20

2pAAd10. Multichannel signal processing at Bell Labs Acoustics Research—Sampled by a postdoc. Walter Kellermann (Multimedia Commun. and Signal Processing, Univ. of Erlangen–Nuremberg, Cauerstr. 7, 91058 Erlangen, Germany)

In the mid 1980's, the first large microphone arrays for audio capture were designed and realized by Jim Flanagan and Gary Elko. After the author joined Bell Labs in 1989, the first real-time digital beamformer for teleconferencing applications was implemented and formed a starting point for the development of several novel beamforming techniques. In parallel, multichannel loudspeaker systems were already investigated and research on acoustic echo cancellation, small-aperture directional microphones, and sensor technology complemented the research scenario aiming at seamless hands-free acoustic communication. Arrays of many sensors and loudspeakers for sampling the spatial domain combined with advanced signal processing sparked new concepts that are still fueling ongoing research around the world—including the author's research group. Here, robust adaptive beamforming has found its way from large-scale arrays into many applications using smaller apertures. Blind source separation algorithms allow for effective spatial filtering without *a priori* information on source positions. Full-duplex communication using multiple channels for both reproduction and recording is enabled by multichannel acoustic echo cancellation combined with beamforming. Recently, wave domain adaptive filtering, a new concept for handling many sensors and many loudspeakers, has been verified for arrays that may well remind some observers of former Bell Labs projects.

4:40

2pAAd11. Room acoustic impulse responses estimation: Challenges and opportunities. Jacob Benesty (Universite du Quebec, INRS-EMT, 800 de la Gauchetiere, Ste. 6900, Montreal, QC H5A 1K6, Canada, benesty@inrs-emt.quebec.ca), Yiteng (Arden) Huang and Jingdong Chen (Bell Labs., Murray Hill, NJ 07974)

The estimation of room acoustic impulse responses in real-time plays a major role in many important applications such as dereverberation, echo and feedback cancellation, noise reduction, etc. This task is very challenging. In this talk, we will discuss different scenarios of estimation, depending on the application. We will see that when this estimation is reliable, the related applications can give good performances. However, in many applications it is very difficult to have a good estimate of the impulse responses. As a result, performances are usually very poor even with very clever tricks. We will see that opportunities are endless if good progress is made in this important area.

Session 2pAB

Animal Bioacoustics: Animal Bioacoustics General Topics

Robert Kull, Chair

5800 Lake Wright Drive, Suite 101, Norfolk, Virginia 23502

Contributed Papers

1:00

2pAB1. Generalized perceptual features for animal vocalization classification. Patrick J. Clemins and Michael T. Johnson (Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881)

Two sets of generalized, perceptual-based features are investigated for use in classifying animal vocalizations. Since many species, especially mammals, share similar physical sound perception mechanisms which vary in size, two features sets commonly used in human speech processing, mel-frequency cepstral coefficients (MFCCs) and perceptual linear prediction (PLP) analysis, are modified for use in other species. One modification made to the feature extraction process is incorporating the frequency range of hearing and length of the basilar membrane of the animal in order to correctly determine the width and location of the critical band filters used for signal processing. Experimentally determined critical bands (equivalent rectangular bandwidth) and equal loudness curves (audiograms) can also be incorporated directly into the feature extraction process. Experiments are performed on African elephant (*Loxodonta africana*) vocalizations using a hidden Markov model (HMM) based classifier showing increased classification accuracy when using features sets based on the specific animals perceptual abilities compared to the original human perception-based feature sets.

1:15

2pAB2. Acoustic and behavioral analyses of gecker distress vocalizations in young rhesus macaques (*Macaca mulatta*). Erik R. Patel and Michael J. Owren (Psych. of Voice and Sound Res. Lab., Dept. of Psych., Cornell Univ., 211 Uris Hall, Ithaca, NY 14853, erp8@cornell.edu)

Loud, pulsed gecker calls have long been known as one of the most common distress vocalizations produced by young rhesus macaques, but have not been systematically investigated. We therefore examined the rates, acoustics, and behavioral contexts associated with geckering, based on audio-recording and focal-animal observations conducted over 4 years at the California National Primate Research Center. The sample analyzed came from 14 young rhesus macaques, with 74 recorded gecker bouts and 556 total pulses. Callers ranged from 1 to 41 months, who in their first year showed most geckering in months 0–2 (26%) and 2–4 (35%). Females geckered longer and at higher rates than males, as well as showing acoustic differences likely to reflect greater vocal effort (e.g., more doubled and tripled pulses, greater noisiness, and higher first frequency peaks). Geckers were associated with 22 differentiable behavioral contexts, typically occurring when with the mother but without obvious antecedents (31.1%), and when the mother broke physical contact with the youngster (13.5%). The most frequent postcedent context of geckering was that most offspring gained the mother's attention and/or renewed proximity (49%). In combination, the outcomes suggest that geckers primarily function to draw listener attention, and are used by youngsters to elicit maternal response.

1:30

2pAB3. Artiodactyl and Perissodactyl acoustics: Identifying distress calls by farm animals. David G. Browning (Dept. of Phys., Univ. of Rhode Island, Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Univ. of Connecticut, Storrs, CT 06269)

There is growing concern for the welfare of farm animals. Vocal signals are discernable in a herd, generally carry over relatively long ranges, and, as Jahns has shown, can be easily automatically detected. Analysis of vocalizations from the two principal farm animal families show, however, that only a few, a pig's squeal, for example, meet Morton's classic criteria for distress. In general, Artiodactyls (cows, sheep, goats, etc.) have tonal bellows or bleats where apparently one vocalization fits many emotional situations. Duration and repetition, as Grandin has suggested, may be the important criteria in indicating stress. In contrast, Perissodactyls vary frequency during some vocalizations, such as a horse whinny, but no direct connection between frequency change and stress has yet been determined. The apparent reliance of Perissodactyles (with keen eyesight) on visual detection of body language appears to limit to some degree the amount of vocalization.

1:45

2pAB4. Vocalization of *Equus asinus*: The hees and haws of donkey brays. David G. Browning (Dept. of Phys., Univ. of Rhode Island, Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Univ. of Connecticut, Storrs, CT 06269)

An understanding of the vocalizations of farm animals may provide an important key in the identification of animals under stress. Donkey vocalizations (brays) appear to be unique among the *Equidae* in that significant sound is produced during both air intake (the hee) and air outflow (the haw). Typically these vocalizations, primarily by males, consist of a series of brays, seemingly mechanically produced with little variation, terminating when the animal becomes short of breath—literally a burst of sound. The acoustic character, duration, and sequence (some hee-haw, while others haw-hee) are unique to each animal. As with other Perissodactyles (horses, zebras), and in contrast to the Artiodactyls (cows, sheep, and goats), there can be a significant frequency variation during a vocalization segment (particularly in the “haw” phase). It has not been determined what control (if any) the animal has over this. Further study is being made of possible braying variations for a given animal.

2:00

2pAB5. Temporal bone anatomy in *Panthera tigris*. Edward J. Walsh (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131), Darlene R. Ketten, Julie Arruda (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Douglas L. Armstrong, Thomas Curro, Lee G. Simmons (Omaha Henry Doorly Zoo, Omaha, NE 68107), Lily M. Wang (Univ. of Nebraska, Omaha, NE 68182), and JoAnn McGee (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

Preliminary findings suggest that members of *Panthera tigris* subspecies may rely on low-frequency acoustic cues when communicating with conspecifics either in the field or in captivity. This view is supported by the observation that individuals are sensitive to tone bursts in the 300–

500-Hz range and produce significant acoustic energy in an overlapping frequency band in the case of close encounter roars. Other utterances within the vocal repertoire of tigers also contain, and are often dominated by, low frequency acoustic energy that can extend into the infrasonic range. Efforts to determine temporal bone correlates of *P. tigris* bioacoustical features were recently initiated using computerized tomography to assess key aspects of middle and inner ear morphology from a small set of adult Siberian tigers (*P. tigris altaica*) and one neonate. Obvious peripheral auditory specializations were not observed and structures comprising the auditory periphery were consistent with the anatomical character of felids generally. Although cochlear dimensions appeared to be adultlike, or nearly so, in the case of the neonate, other temporal bone features were grossly immature. The relationship between acoustic sensitivity, the spectral character of a subset of close encounter calls and cochlear dimensions will be considered.

2:15

2pAB6. Sound detection and production in the American lobster, *Homarus americanus*: Sensitivity range and behavioral implications. Heidi J. Pye and Winsor H. Watson III (Univ. of New Hampshire, 46 College Rd., Durham, NH 03824)

Many crustaceans including spiny lobsters, some crabs, and a few shrimp are known to produce sounds for a variety of purposes. One brief report and several preliminary studies indicate that American lobsters also produce sounds, and may be capable of detecting acoustic signals. The focus of this study was to (1) quantify the frequency range over which lobsters are capable of detecting sounds and (2) characterize the sounds that lobsters produce. Twelve sexually immature and 11 mature lobsters were tested for their ability to detect frequencies in the range of 20–10 000 Hz. Immature lobsters of both sexes detected sounds in the range of 20–1000 Hz (>50%), while sexually mature lobsters exhibited two distinct peaks in their acoustic sensitivity (20–300 Hz and 1000–5000 Hz). Lobsters of both sexes produced a buzzing vibration when grasped but larger lobsters (120–149 mm in carapace length) vibrated most consistently (>35% of surveyed lobsters). The greater tendency for sound production in large lobsters may indicate a role in mating behavior. Currently, we are characterizing the acoustical properties of produced sounds and investigating the possibility that the American lobster may produce sounds for more than one purpose. [Work supported by University of New Hampshire Center for Marine Biology.]

2:30

2pAB7. Probability density functions for hyperbolic and isodiachronic locations. John L. Spiesberger (Dept. of Earth and Environ. Sci., Univ. Pennsylvania, 240 S. 33rd St., Philadelphia, PA 19104, johnsr@sas.upenn.edu) and Magnus Wahlberg (Aarhus Univ., DK-8000 Aarhus C, Denmark)

Animal locations are sometimes estimated with hyperbolic techniques by estimating the difference in distances of their sounds between pairs of receivers. Each pair specifies the animal's location to a hyperboloid because the speed of sound is assumed to be spatially homogeneous. Sufficient numbers of intersecting hyperboloids specify the location. A non-linear method is developed for computing probability density functions for location. The method incorporates *a priori* probability density functions for the receiver locations, the speed of sound, winds, and the errors in the differences in travel time. The traditional linear approximation method overestimates bounds for probability density functions by one or two orders of magnitude compared with the more accurate non-linear method. The non-linear method incorporates a generalization of hyperbolic methods because the average speed of sound is allowed to vary between different receivers and the source. The resulting "isodiachronic" surface is the locus of points on which the difference in travel time is constant. Isodiachronic locations yield correct location errors in situations where hyperbolic methods yield incorrect results, particularly when the speed of propagation varies significantly between a source and different receivers.

3:00

2pAB8. Information entropy analysis of leopard seal vocalization bouts. John R. Buck (ECE Dept. and SMAST, UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747), Tracey L. Rogers (Zoological Parks Board of NSW, Mosman, NSW 2088, Australia), and Douglas H. Cato (Defence Sci. and Technol. Organisation, Pyrmont, NSW 2009, Australia)

Leopard seals (*Hydrurga leptonyx*) are solitary pinnipeds who are vocally active during their brief breeding season. The seals produce vocal bouts consisting of a sequence of distinct sounds, with an average length of roughly ten sounds. The sequential structure of the bouts is thought to be individually distinctive. Bouts recorded from five leopard seals during 1992–1994 were analyzed using information theory. The first-order Markov model entropy estimates were substantially smaller than the independent, identically distributed model entropy estimates for all five seals, indicative of constraints on the sequential structure of each seal's bouts. Each bout in the data set was classified using maximum-likelihood estimates from the first-order Markov model for each seal. This technique correctly classified 85% of the bouts, comparable to results in Rogers and Cato [Behaviour (2002)]. The relative entropies between the Markov models were found to be infinite in 18/20 possible cross-comparisons, indicating there is no probability of misclassifying the bouts in these 18 comparisons in the limit of long data sequences. One seal has sufficient data to compare a nonparametric entropy estimate with the Markov entropy estimate, finding only a small difference. This suggests that the first-order Markov model captures almost all the sequential structure in this seal's bouts.

3:15

2pAB9. Localization of airborne pure tones by pinnipeds. Marla M. Holt, Ronald J. Schusterman, Brandon L. Southall, and David Kastak (Univ. of California, Santa Cruz Long Marine Lab., 100 Shaffer Rd., Santa Cruz, CA 95060, iris@ucsc.edu)

Although all pinnipeds communicate acoustically in air, most previous research on sound localization has been done under water. We have recently shown that several pinniped species localize aerial broadband signals as well as some terrestrial carnivores [Holt *et al.*, J. Acoust. Soc. Am. **113** (2003)]. However, it is unclear which frequencies are particularly important for localization in these animals. In this study, we tested a harbor seal (*Phoca vitulina*) and a California sea lion (*Zalophus californianus*) in a hemianechoic chamber at frequencies ranging between 0.8 and 20 kHz. A left/right procedure was used to measure minimum audible angles (MAAs) corresponding to 75%-correct discrimination. MAAs ranged from approximately 4 to 13 deg in both subjects, with the largest MAAs or poorest acuity measured at the intermediate frequencies tested. These results are consistent with the duplex theory of sound localization in that low-frequency sounds appear to be localized on the basis of interaural time differences, while high-frequency sounds appear to be localized on the basis of interaural intensity differences. Testing with a northern elephant seal (*Mirounga angustirostris*) will provide further insight on the use of binaural cues and head-size effects with respect to localization in pinnipeds.

3:30

2pAB10. Determination of West Indian manatee vocalization levels and rate. Richard Phillips, Christopher Niezrecki, and Diedrich Beusse (Univ. of Florida, Gainesville, FL 32611-6250, niezreck@ufl.edu)

The West Indian manatee (*Trichechus manatus latirostris*) has become endangered partly because of a growing number of collisions with boats. A system to warn boaters of the presence of manatees, based upon the vocalizations of manatees, could potentially reduce these boat collisions. The feasibility of this warning system would depend mainly upon two factors:

the rate at which manatees vocalize and the distance in which the manatees can be detected. The research presented in this paper verifies that the average vocalization rate of the West Indian manatee is approximately one to two times per 5-min period. Several different manatee vocalization recordings were broadcast to the manatees and their response was observed. It was found that during the broadcast periods, the vocalization rates for the manatees increased substantially when compared with the average vocalization rates during nonbroadcast periods. An array of four hydrophones was used while recording the manatees. This allowed for position estimation techniques to be used to determine the location of the vocalizing manatee. Knowing the position of the manatee, the source level was determined and it was found that the mean source level of the manatee vocalizations is approximately 112 dB (*re*:1 Pa) @ 1 m.

3:45

2pAB11. Measurement of Lombard-like response in the beluga whale. Peter M. Scheifele (Animal Sci. Dept., Univ. of Connecticut, 3636 Horsebarn Hill Rd. Ext., Unit 4040, Storrs, CT 06269-4040, scheifele@uconn.edu)

Noise pollution has become recognized as a potential danger to marine mammals in general, and to the St. Lawrence beluga (*Delphinapterus leucas*) in particular. One method to determine whether noise is having an effect on an animal's auditory ability is to observe a natural and repeatable response of the auditory and vocal systems to varying noise levels. This can be accomplished by observing changes in animal vocalizations in response to auditory feedback. A response such as this observed in humans and some animals is known as the Lombard vocal response, which represents a reaction of the auditory system directly manifested by changes in vocalization level. This response is known in humans, songbirds, and some primates. In this research a population of belugas in the St. Lawrence River Estuary was tested to determine whether a vocalization-as-a-function-of-noise phenomenon existed by using hidden Markov classified vocalizations as targets for acoustical analyses. Correlation and regression analyses of signals and noise indicated that the phenomenon does exist and results of a human subjects experiment along with results from other animal species known to exhibit the response strongly implicate the Lombard vocal response in the St. Lawrence population of beluga.

4:00

2pAB12. Comparative sensitivity analysis of transmission loss in beaked whale environments. Eryn M. Wezensky, James H. Miller, and Robert C. Tyce (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882)

Scientific literature states that anthropogenic sound, such as mid-frequency sonar, may cause a behavioral response in marine mammals. The degree of response is highly variable and dependent upon many factors, including how sound transmission is influenced by environmental features. The physical parameters of the ocean medium, such as sound speed profile and bathymetry, are important controls of underwater acoustic propagation. Determining the acoustic propagation loss of the ocean environment is an application used to identify and correlate influential environmental factors. This study investigates the sensitivity of acoustic propagation loss based on specific physical characteristics found in five different sites representing beaked whale environments. These sites were chosen with regards to existing data on beaked whale distribution, historical mass stranding records, and presence of mid-frequency sonar activity. A range-independent, ray-tracing acoustic propagation model was used to generate a two-dimensional sound field over a range of 30 km. From the results of this experiment, the acoustic importance of bathymetry and sound speed profile of the five beaked whale environments were identified. Preliminary results from the experimental study will be presented.

4:15

2pAB13. Detecting sperm whale clicks in the presence of ambient and shipping noise using higher order moments. James P. Larue (AFRL/IFEC, Rome, NY 13441), George E. Ioup, and Juliette W. Ioup (Univ. of New Orleans, New Orleans, LA 70148)

The single-receiver detection of various sources using higher order moments is demonstrated. The data for this study come from the Littoral Acoustic Demonstration Center experiment in the northern Gulf of Mexico in the summer of 2001. Results show that in a time-varying environment it may be more meaningful to use non-normalized statistics (e.g., using the fourth central moment rather than the kurtosis). A whale click is detected over a 35-ms window, i.e., with a sampling rate of 11 718 samples per second, 40 samples are used to produce a statistic. In the past, energy and related detectors have been used for the single receiver detection of broadband signals produced by whale clicks. The presence of loud shipping noise, which is also broadband, tends to mask the clicks and make their detection difficult. Using short-time statistics may help mitigate this problem. Analysis with fractal exponents will be shown as well. [Research supported by ONR.]

2p TUE. PM

Session 2pBB**Biomedical Ultrasound/Bioresponse to Vibration and Physical Acoustics: High Intensity Focused Ultrasound II**

Cheri Deng, Cochair

Case Western Reserve University, 10900 Euclid Avenue, Cleveland, Ohio 44106

Shahram Vaezy, Cochair

*University of Washington, HSB Box 356540, Seattle, Washington 98195-6540***Chair's Introduction—1:00*****Invited Papers*****1:05**

2pBB1. Acoustic hemostasis. Lawrence Crum, Marilee Andrew, Michael Bailey, Kirk Beach, Peter Kaczkowski, Roy Martin, Shahram Vaezy (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), and Vera Khokhlova (Moscow State Univ., Moscow 119992, Russia)

The application of HIFU to tissue can result in rapid temperature elevations, so much so that protein denaturation can occur. Not only can these elevated temperatures and denatured proteins induce coagulative necrosis, but they can also result in the cessation of blood supply to a particular region. Indeed, when applied to a site of active bleeding, hemostasis can be induced. A group of us at the University of Washington have considered the application of HIFU to active bleeding and have developed methods and devices that offer considerable promise for use in clinical medicine. A brief review of our work in this area will be presented, including our most recent results. [Work supported in part by the U.S. Army and the NIH.]

1:25

2pBB2. Monitoring of HIFU-induced lesions using active imaging methods. Hui Yao and Emad Ebbini (Dept. of Elec. and Computer Eng., Univ. of Minnesota, Minneapolis, MN 55455)

The use of active imaging methods in monitoring the properties of HIFU-induced lesions in freshly excised tissue is investigated. A modern ultrasound scanner is used to collect high-frame rate ultrasound images in conjunction with short-duration localized tissue modification using an external focused transducer. Two modes of reversible tissue modification are used: mechanically pushing the tissue using millisecond pulses and incrementally heating the tissue using 0.5–1 s duration pulses. Speckle tracking techniques are used to estimate the tissue displacements due to shear wave generation (in the case of ms pulse excitation) and temperature change (in the case of 1 s pulse excitation). Experimental results have shown that 20%–25% change in the shear elastic modulus occurs after lesion formation and appears to be persistent up to 20 min after lesion formation. Absorption appears to increase significantly immediately after heating, but drops nearly to prelesion levels 5 min after heating. A description of the real-time ultrasonic monitoring system will be given. The principles of measurement of both thermal and viscoelastic properties are also described. Illustrative experimental examples from tissue data are given and discussed.

1:45

2pBB3. Fast algorithm for nonlinear acoustics and high-intensity focused ultrasound modeling. Francesco P. Curra, Steven G. Kargl, and Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA, fcurra@apl.washington.edu)

The inhomogeneous characteristics of biological media and the nonlinear nature of sound propagation at high-intensity focused ultrasound (HIFU) regimes make accurate modeling of real HIFU applications a challenging task in terms of computational time and resources. A fast, dynamically adaptive time-domain method that drastically reduces these pitfalls is presented for the solution of multidimensional HIFU problems in complex geometries. The model, based on lifted interpolating second-generation wavelets in a collocation approach, consists of the coupled solution of the full-wave nonlinear equation of sound with the bioheat equation for temperature computation. It accounts for nonlinear acoustic propagation, arbitrary frequency power law for attenuation, multiple reflections, and backscattered fields. The characteristic localization of wavelets in both space and wave number domains allows for accurate simulations of strong material inhomogeneities and steep nonlinear processes at a reduced number of collocation points, while the natural multiresolution analysis of wavelets decomposition introduces automatic grid refinement in regions where localized structures are present. Compared to standard finite-difference or spectral schemes on uniform fine grids, this method shows significant savings in computational time and memory requirements proportional with the dimensionality of the problem. [Work supported by U.S. Army Medical Research Acquisition Activity through the University.]

2pBB4. Interstitial ablation and imaging of soft tissue using miniaturized ultrasound arrays. Inder R. S. Makin, Laura A. Gallagher, T. Douglas Mast, Megan M. Runk, Waseem Faidi (Ethicon Endo-Surgery, 4545 Creek Rd., Cincinnati, OH 45242, imakin@eesus.jnj.com), Peter G. Barthe, and Michael H. Slayton (Guided Therapy Systems, Mesa, AZ 85202)

A potential alternative to extracorporeal, noninvasive HIFU therapy is minimally invasive, interstitial ultrasound ablation that can be performed laparoscopically or percutaneously. Research in this area at Guided Therapy Systems and Ethicon Endo-Surgery has included development of miniaturized (~3 mm diameter) linear ultrasound arrays capable of high power for bulk tissue ablation as well as broad bandwidth for imaging. An integrated control system allows therapy planning and automated treatment guided by real-time interstitial B-scan imaging. Image quality, challenging because of limited probe dimensions and channel count, is aided by signal processing techniques that improve image definition and contrast. Simulations of ultrasonic heat deposition, bio-heat transfer, and tissue modification provide understanding and guidance for development of treatment strategies. Results from *in vitro* and *in vivo* ablation experiments, together with corresponding simulations, will be described. Using methods of rotational scanning, this approach is shown to be capable of clinically relevant ablation rates and volumes.

2pBB5. Clinical exposure protocols for HIFU treatments of cancer of the liver and kidney. Gail R. ter Haar (Phys. Dept., Royal Marsden Hospital, Sutton, Surrey SM2 5PT, UK), Ian H. Rivens (Royal Marsden Hospital, Surrey SM2 5PT, UK), James E. Kennedy (Churchill Hospital, Oxford OX3 7LJ, UK), and Feng Wu (Chongqing Univ. of Medical Sci., Chongqing, PRC)

High-intensity focused ultrasound surgery (HIFU) is a technique that is finding increasingly widespread clinical application. The potential of HIFU for the noninvasive treatment of deep-seated tumors is being explored. Treatment of these tumors uses an extracorporeal approach. A number of different devices are now available for delivering clinical HIFU treatments. This paper sets out methods for characterization of HIFU transducers and exposure protocols that would allow treatments to be compared. Clinical trials for the treatment of tumors of the liver and kidney are underway in the UK at the Royal Marsden Hospital, London and the Churchill Hospital, Oxford. This paper describes the physical and technical characteristics of the two devices being used to deliver treatments, and uses these as a basis for discussing the ultrasonic procedures that can be used, and the rationale for their choice. Clinical reference is given for illustrative purposes. The wide variety of exposure protocols that is being used in different clinical centers means that there is at present no uniformity of reporting in the literature. This paper therefore sets out recommended information that should be included in order that transducers and treatments can be compared.

2pBB6. Prostate cancer treated with HIFU: A 10-year experience. Jean-Yves Chapelon, Laura Curiel, and Albert Gelet (INSERM U556, 151, Cours Albert Thomas, 69424 Lyon, Cedex 03, France, chapelon@lyon.inserm.fr)

Objectives: To evaluate our overall experience in HIFU in the prostate cancer application, for its two main indications: as a primary care and as a salvage therapy after radiation. Material and method: Group 1: patients treated with HIFU as primary care for localized prostate cancer, with a baseline PSA level lower than 30 ng/ml, and with at least 1 year of follow-up were included in this analysis. Group 2: patients with a local recurrence after radiation. They were all treated using the Ablatherm HIFU device (EDAP S.A., France). Results: Group 1: $n = 242$ patients. The mean nadir PSA was 0.63 sd 1.30 ng/ml, and further follow-up evidenced a 81% negative biopsy rate. These results are influenced by the usual prognostic factors (clinical stage, baseline PSA, Gleason score), and, as for the other treatment option, the nadir allows prediction of the patient outcome. Group 2: $n = 71$ patients. After HIFU treatment, 80% of the patients presented negative biopsies, and 61% had a nadir PSA level lower than 0.5 ng/ml, obtained within 3 months. No rectal injury occurred since the use of specific parameters. Conclusion: Transrectal HIFU is a valuable option for prostate cancer, for primary care as well as for salvage therapy.

Contributed Papers

2pBB7. Real-time motion correction combined with electronic beam steering strategies optimized for large tumors: First *in-vitro* HIFU experiments in large moving targets. Mathieu Pernot, Mickael Tanter, and Mathias Fink (Laboratoire Ondes et Acoustique, E.S.P.C.I., 10 rue Vauquelin 75231 Paris Cedex 05, France, mathieu.pernot@loa.espci.fr)

A method for tracking in realtime the 3D motion of tissues is combined with a 2D high-intensity focused ultrasound (HIFU) multichannel system in order to correct for respiratory motion during HIFU therapy. Displacements estimation is based on an accurate 3D ultrasonic speckle tracking technique. A correction is achieved in real time by adjusting the transmit delays of each channel of the HIFU system. *In vitro* HIFU ex-

periments combined with motion correction are performed in fresh biological tissues. The accuracy of the HIFU targeting is clearly improved. Moreover, this technique permits an important reduction of the treatment time for moving tumors. To generate large thermal lesions, the focus is scanned electronically over a large region. Different heating strategies are investigated in moving tissues: the shot-by-shot treatment interleaved with cooling periods, and the continuously scanned treatment of the whole region. The temperature distribution and the thermal dose are computed using a 3D thermal diffusion code combined with a 3D wave diffraction algorithm and compared to experimental results. Using motion correction, the necrosis threshold is achieved 3 times faster than without motion correction. This should lead to an important time reduction for the treatment of abdominal tumors.

2pBB8. Therapy/imaging array-based system and technology for intense ultrasound surgery. Peter G. Barthe, Michael H. Slayton, Paul M. Jaeger (Guided Therapy Systems, 33 S. Sycamore St., Mesa, AZ 85202, p.barthe@guidedtherapy.com), Inder R. S. Makin (Ethicon Endo-Surgery, Cincinnati, OH 45242), Laura A. Gallagher, T. Douglas Mast, Megan M. Runk, and Waseem Faidi (Ethicon Endo-Surgery, Cincinnati, OH 45242)

Minimally invasive, miniature (2.2- × 50-mm aperture, 3.3-mm diameter) dual-mode linear arrays have been developed into low-cost disposable probes with high acoustic power output (120 W/cm² at the source), high transmit efficiency (>65% typical), and good imaging performance (50% fractional bandwidth, >100-mm-deep field of view). These therapy/imaging probes have been integrated into a flexible intense ultrasound surgery platform which also includes conventional diagnostic imaging probes. A system architecture has been developed which includes a 64-channel therapy driver with software selection of array aperture and phasing ($\lambda/16$), frequency (0.5–8 MHz), drive amplitude (5 W/channel, nominal), rotational steering (± 180 deg), and temporal sequencing/switching of imaging/therapy/monitoring modes. System software includes graphical and text-based script mode control of therapeutic treatment. Real-time monitoring of electric power per channel, temperature sensors, and thermal effects provide a range of feedback and safety. Numerous system and probe technological issues such as electrical interconnect and matching, acoustic coupling, thermal control, and maintaining probe efficiency have been addressed. The array-based imaging/therapy system has produced encouraging results in preclinical studies of bulk tissue ablation and imaging.

3:50

2pBB9. Dual-modality phased array systems for hyperthermia cancer therapy. Robert McGough (Dept. of Elec. and Computer Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824) and Thaddeus Samulski (Duke Univ. Medical Ctr., Durham, NC 27710)

External ultrasound (US) phased arrays, as applied to hyperthermia cancer therapy, are generally local heating devices, whereas external radio-frequency (RF) electromagnetic (EM) phased array applicators are primarily designed for regional heating. Clinical applications of external ultrasound heating devices are presently limited to smaller tumors, and the inclusion of an external RF/EM applicator increases the size of the heated volume. When these two modalities are combined in a hybrid RF/US phased array structure, uniform heating is facilitated in larger tumors. This is demonstrated in computer simulations of thermal therapy in the breast, where adding an external RF/EM component to an ultrasound phased array in a hybrid structure produces temperatures of 43 °C in nearly half of a 9-cm-diam tumor volume. If applied alone, neither modality achieves 43 °C in such a large volume without overheating intervening tissues. This result is demonstrated in two hybrid RF/US applicator prototypes designed for hyperthermia treatments of locally advanced breast cancer. In each example, hybrid RF/US phased arrays achieve higher temperatures in larger tumor volumes by selectively delivering power to different regions in the tumor with each modality.

4:05

2pBB10. A multi-element high intensity focused ultrasound transducer: Design, fabrication, and testing. Shahram Vaezy, Robert Held, Blake Miller (Appl. Phys. Lab., Univ. of Washington, Box 355640, vaezy@apl.washington.edu), and Gerard Fleury (Imasonic Corp., Besancon, France)

The goal of this project is to develop an intra-cavity image-guided high intensity focused ultrasound (HIFU) device using piezocomposite technology and commercially available ultrasound imaging. The HIFU array, manufactured by Imasonic Corporation, is an 11-element annular phased array, with a focal length range of 30–60 mm, and operating frequency of 3 MHz (bandwidth of 1 MHz). The imaging probe (C9-5, Philips) is configured such that the focal axis of the HIFU beam was within the image plane. The array includes six complete central rings and five side-

truncated peripheral rings, all with the natural radius of curvature of 50 mm. Impedance of all elements is approximately 50 ohms (10% accuracy for real and imaginary parts). Cross coupling between adjacent elements is less than, –40 dB. High power measurements showed more than 75% efficiency, at surface intensity of 2.66 W/cm². Schlieren imaging showed effective focusing at all focal lengths (30–60 mm). The image-guided HIFU device requires water or hydrogel coupling, and possibly water cooling. The results of the full characterization for lesion formation in tissue-mimicking phantoms and biological tissues will be presented. Possible applications include uterine fibroids, abnormal uterine bleeding, and intraoperative hemostasis of occult hemorrhage.

4:20

2pBB11. High-intensity focused ultrasound for liver biopsy hemostasis. Cheri Deng, Hesheng Wang, Yun Zhou (Dept. of Biomed. Eng., Case Western Reserve Univ., 10900 Euclid Ave., Cleveland, OH 44106-7207, cxd54@cwru.edu), Vikram Dogra, Agata Exner, Shweta Bhatt, John Haaga, and Nicholas Stowe (Univ. Hospitals of Cleveland, Cleveland, OH 44106)

In vivo experiments were conducted to demonstrate the feasibility of HIFU application to control postliver biopsy hemorrhage. Yorkshire pigs were anesthetized and their livers were surgically exposed. Core biopsies ($n=74$) were performed on the exposed hepatic parenchyma with 14-gauge ($n=41$) and 18-gauge ($n=33$) core biopsy needles that were inserted 1.5–2 cm deep into the liver. Hemorrhage was determined from the weight of the blood collected from each biopsy puncture site using surgical sponges immediately after biopsy needle retraction. To stop hemorrhage, immediate HIFU was applied to the needle entry site ($n=44$) after needle retraction. HIFU was generated using a piezoelectric (PZT) transducer (diameter=42 mm, F number=1.2) at 4.23 MHz. Whole-blood clotting times were measured at various times throughout the experiments. Mean blood loss from control biopsy sites using a 14-gauge needle ($n=18$) was 1.78 g, while mean blood loss using an 18-gauge needle ($n=10$) was 1.22 g (two 14-gauge-needle control biopsies were excluded). Virtually no blood loss was measured from the biopsy needle entry site after HIFU application for both 14- and 18-gauge-needle biopsies. Ultrasound imaging demonstrated a marked difference between control sites and HIFU-treated sites where successful hemostasis was achieved.

4:35

2pBB12. A model-based noninvasive temperature estimation technique for monitoring HIFU therapy using backscattered ultrasound. Ajay Anand and Peter J. Kaczkowski (Ctr. for Industrial and Med. Ultrasound, APL, Univ. of Washington, 1013 40th St. NE, Seattle, WA 98105, ajaya@apl.washington.edu)

A new noninvasive temperature estimation technique using the bioheat transfer equation (BHTE) to constrain inversion of ultrasonic travel time data has been developed for HIFU therapy monitoring. Initial estimates for BHTE parameters and ultrasonic travel time temperature dependence $\delta\tau(T)$ are noninvasively obtained *in situ* using ultrasound backscatter signals acquired during two probe heating exposures performed prior to therapy: one to obtain medium thermal parameters, the other to estimate the magnitude of the local HIFU heat source. These estimates are used in BHTE simulations and compared to local travel time changes measured periodically during therapy using a medical ultrasound imager. Quantitative 2-D temperature maps are obtained by constrained data fitting. Preliminary results obtained from *in vitro* experiments in tissue mimicking phantoms will be presented. Temperature maps are computed throughout therapy delivery and posttreatment cooling periods, and validated against independent temperature measurements obtained using thermocouples placed close to but not at the HIFU focus. This model-based technique permits temperature estimation throughout the entire therapeutic range and is completely noninvasive, and is thus a departure from previously reported techniques. This ultrasound-based approach could serve as an alternative to MRI-based temperature estimation methods in a clinical setting.

4:50

2pBB13. Dosimetry measurement of HIFU field using a fiber-optical probe hydrophone. Liang Zhai, Yufeng Zhou, and Pei Zhong (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Box 90300, Durham, NC 27708, pzhong@duke.edu)

Characterization of the acoustic field of high-intensity focused ultrasound (HIFU) transducers by conventional PVDF membrane and needle hydrophones is problematic due to limited bandwidth, spatial averaging, and damage to the hydrophone. Here, we report the use of a self-calibrated fiber-optical probe hydrophone (FOPH-500) for HIFU dosimetry measure-

ment. The hydrophone (0.1-mm sensing element) was scanned in the focal volume of a 1.1-MHz HIFU transducer ($F=63$ mm, f number=0.9) at 0.2-mm steps using a computer-controlled 3D positioning system. When the input voltage V_{p-p} applied to the transducer was increased from 28 to 225 volts, the peak compressive and tensile pressure values at the transducer focus were found to be $P^+=1.7\text{--}18.9$ MPa and $P^-=-1.33\text{--}-8.14$ MPa, respectively. The corresponding spatial peak intensities were calculated to be $I_{sp}=69\text{--}3968$ W/cm². Nonlinear propagation with harmonics generation was dominant at high intensity levels, leading to a reduced -6-dB beamwidth of the compressive wave from 1.8 to 1.3 mm and an increased -6-dB beamwidth of the tensile wave from 1.6 to 1.8 mm. Overall, FOPH-500 was found to be a reliable tool for characterizing the acoustic field of HIFU transducers.

TUESDAY AFTERNOON, 25 MAY 2004

LIBERTY 5, 1:30 TO 4:00 P.M.

Session 2pED

Education in Acoustics: Apparatus for Teaching Acoustics, 1929 and Before

Thomas D. Rossing, Cochair

Physics Department, Northern Illinois University, De Kalb, Illinois 60115

Peter L. Hoekje, Cochair

Department of Physics and Astronomy, Baldwin-Wallace College, 275 Eastland Road, Berea, Ohio 44017

Chair's Introduction—1:30

Invited Papers

1:35

2pED1. Apparatus for studying wave motion and sound at the University of Nebraska—Lincoln's "Historical Scientific Instrument Gallery." Lily M. Wang (Architectural Eng. Prog., Univ. of Nebraska, Peter Kiewit Inst., Omaha, NE 68182-0681, lwang4@unl.edu) and M. Eugene Rudd (Univ. of Nebraska, Lincoln, NE 68588)

The University of Nebraska—Lincoln's "Historical Scientific Instrument Gallery," compiled by the second author in 1998, contains approximately 700 inventoried items and may be visited on-line at <http://physics.unl.edu/outreach/histinstr/>. Amidst the collection are several acoustical instruments that were used in the early 1900s. These include equipment that demonstrate wave motion (traveling wave machine, mercury ripple dish, vibration microscope), wave interference (interference machine), resonance conditions (Helmholtz resonators, vibrating rods, singing flames, sonometer), and sound generation (Galton's whistles, high-frequency tuning forks, large tuning forks, organ pipes, siren saw). A review of the equipment and the history of their use at the University of Nebraska are discussed. Much of the equipment was superbly manufactured by the Max Kohl/Chemnitz Company in Germany and Rudolph Koenig in France. Pages from the Max Kohl/Chemnitz equipment catalogs of 1910 and 1925 helped to characterize several of the pieces and are shown in this presentation.

1:55

2pED2. Early 20th century acoustics apparatus in Iowa. Roger J. Hanson (Dept. of Phys., Univ. of Northern Iowa, Cedar Falls, IA 50613, roger.hanson@cfu.net)

In the first half of the 20th century G. W. Stewart was a physics faculty member at the University of Iowa (UI) with a distinguished record of research and teaching, especially in acoustics. Much of his research focused on the design and use of several types of acoustical filters. Some apparatus which he developed or utilized are still housed in the Department of Physics and Astronomy or are available in detailed diagrams. Demonstration apparatus (apparently homemade) from his era are still available for use. Carl E. Seashore, a renowned psychologist also at UI in the early 20th century, had interdisciplinary interests linking psychology, speech and hearing, music, and acoustics. He was responsible for obtaining an Henrici harmonic analyzer, a mechanical Fourier analyzer manufactured in Switzerland, a special grant from the state legislature during Depression conditions provided the funding. It resides in the Department of Speech Pathology and Audiology at UI. The Grinnell College Physics Historical Museum houses a set of 18 Helmholtz resonators and a Savart bell and resonator. Apparatus at Iowa State University, the University of Northern Iowa, and other Iowa institutions will also be described. Pictures and diagrams as well as some actual apparatus will be exhibited.

2pED3. Rudolph Koenig's workshop of sound. David A. Pantalony (Dibner Inst., MIT, 38 Memorial Dr., Cambridge, MA 02139, pantalony@mit.edu)

Rudolph Koenig's workshop was a busy meeting place for instruments, ideas, experiments, demonstrations, craft traditions, and business. Starting around 1860 it was also the place in Paris where people discovered the new science of sound emerging from the studies of Hermann von Helmholtz in Germany. Koenig built Helmholtz's ideas into apparatus, created new instruments, and spread them throughout the scientific and musical world. Through his own research, he also became Helmholtz's strongest critic. This paper looks at the activities of this unique space, and, in particular, how it contributed to the protracted disputes over an elusive acoustical phenomenon called the combination tone. Many of these instruments became standard teaching and demonstration apparatus.

2pED4. Harmonic analysis utilizing a Phonodeik and an Henrici analyzer. William J. Fickinger (Dept. of Phys., Case Western Reserve Univ., Cleveland, OH 44106, wjf@case.edu), Roger J. Hanson (Univ. of Northern Iowa, Cedar Falls, IA 50614), and Peter L. Hoekje (Baldwin-Wallace College, Berea, OH 44017)

Dayton C. Miller of the Case School of Applied Science assembled a series of instruments for accurate analysis of sound [D. C. Miller, *J. Franklin Inst.* **182**, 285–322 (1916)]. He created the Phonodeik to display and record sound waveforms of musical instruments, voices, fog horns, and so on. Waveforms were analyzed with the Henrici harmonic analyzer, built in Switzerland by G. Coradi. In this device, the motion of a stylus along the curve to be analyzed causes a series of spheres to rotate; two moveable rollers in contact with the n th sphere record the contributions of the $\sin(nx)$ and $\cos(nx)$ components of the wave. Corrections for the measured spectra are calculated from analysis of the response of the Phonodeik. Finally, the original waveform could be reconstructed from the corrected spectral amplitudes and phases by a waveform synthesizer, also built at Case. Videos will be presented that show the motion of the gears, spheres, and dials of a working Henrici analyzer, housed at the Department of Speech Pathology and Audiology at the University of Iowa. Operation of the Henrici analyzer and the waveform synthesizer will be explained.

2pED5. Acoustic teaching apparatus before 1929 at the Case School of Applied Science. Peter L. Hoekje (Dept. of Phys. and Astron., Baldwin-Wallace College, 275 Eastland Rd., Berea, OH 44017, phoekje@bw.edu) and William Fickinger (Case Western Reserve Univ., Cleveland, OH 44106)

The acoustics apparatus found in the Physics Department of the Case School of Applied Science in the first decades of the 20th century included many items common to other acoustical teaching laboratories, such as organ pipes, tuning forks, Helmholtz resonators, sirens, and manometric flame sound analyzers. The European instrument makers Rudolf Koenig and Max Kohl supplied much of this. Equipment built at Case included the phonodeik, which Dayton C. Miller designed in 1908, and the waveform synthesizer. Miller supplied detailed descriptions of the operations of all this equipment in papers and books. In the phonodeik (to show sound), sound deflects a thin glass diaphragm, which by a silk thread turns a mirror on an axle, causing a spot of light to move across film or a projection screen. A working model of the phonodeik has been reconstructed from pieces of two original ones, and will be demonstrated. Photographs of other extant instruments in the collection, and a selection from Millers lantern slides, will be displayed.

Contributed Papers

2pED6. A model for precalculus students to determine the resonance frequency of a trumpet mouthpiece. Robert C. Chapman (Jefferson County Public Schools, 1829 Denver West Dr., Golden, CO 80401-3120)

The trumpet mouthpiece as a Helmholtz resonator is used to show precalculus students a mathematical model for determining the approximate resonance frequency of the mouthpiece. The mathematics is limited to algebra and trigonometry. Using a system of mouthpieces that have interchangeable cups and backbores, students are introduced to the acoustics of this resonator. By gathering data on 51 different configurations of mouthpieces, the author modifies the existing Helmholtz resonator equation to account for both cup volumes and backbore configurations. Students then use this model for frequency predictions. Included are how to measure the different physical attributes of a trumpet mouthpiece at minimal cost. This includes methods for measuring cup volume, backbore

volume, backbore length, throat area, etc. A portion of this phase is designed for students to become acquainted with some of the vocabulary of acoustics and the physics of sound.

2pED7. Acoustic engineering at Universidad de las Americas, Ecuador. Luis A. Bravo, Jaime O. Naranjo, and Alberto Tassara (Coln 338 y 6 de Diciembre, Quito, Ecuador, lbravo@uamericas.edu.ec)

Acoustics, like science, an instrument to develop new technologies, comfortable atmospheres, and pleasant sounds, has not had a sufficient push in Ecuador. The shortage of professionals in the area, and the social ignorance of the advances and benefits of acoustics have been part of the problem. The University of the Americas has taken the initiative to develop an undergraduate program—only in the country—of sound and acoustics engineering, to contribute to the formation of professional fu-

tures that fortify the recent labor market in the areas of audio, professional, and acoustic engineering. This work presents/displays the results of the studies made for the creation of the race, the curricular mesh, and its projections.

3:45

2pED8. Comparison of Russian and American educational programs in acoustics. Nikolai Zagrai and Vladislav Zakharevich (Taganrog State Univ. of Radio-Eng., GSP-17A, 44 Nekrasovsky St., Taganrog 347928, Russia, znp@tsure.ru)

Recent progress in finding parallels between Russian and American professional education programs has shown that particular attention should be paid to the differences and similarities in engineering education. The

efforts of the Taganrog State University of Radio-Engineering (TSURE) are directed towards accreditation of its engineering programs in ABET. The Russian Bachelor program in acoustics was compared to the ABET-accredited American program in electrical engineering. A direct comparison has shown that Russian program has: (a) more credit hours in the course work under the liberal arts section (factor of 2); (b) traditionally more credit hours in mathematics and fundamental science (factor of 1.5); (c) same number of credit hours for the general engineering classes; (d) less number of credit hours for the specialized engineering classes. However, by examining the actual class contents of the specialized engineering classes in the American program, we observed that some classes contain material which, in Russian program, is usually attributed to the general engineering. Recent revision of the acoustic education program at TSURE aimed to increase its flexibility, support students' mobility, and facilitate ABET accreditation.

TUESDAY AFTERNOON, 25 MAY 2004

CONFERENCE ROOM K, 1:00 TO 3:20 P.M.

Session 2pMU

Musical Acoustics and Signal Processing in Acoustics: Digital Signal Processing Methods for Restoring, Enhancing and Manipulating Music Recordings

James W. Beauchamp, Chair

School of Music, Department of Electrical and Computer Engineering, University of Illinois—Urbana/Champaign, 1114 West Nevada, Urbana, Illinois 61801-3859

Invited Papers

1:00

2pMU1. Retrieving the sources in historical sound recordings. George Brock-Nannestad (Patent Tactics, Resedavej 40, DK-2820 Gentofte, Denmark, pattac@image.dk)

Broadly speaking, historical recordings (ca. 1880–1950) are considered low quality, and when they are reissued it is regarded as an improvement when noises are removed, or according to modern psychoacoustic criteria, pushed into frequency bands where they are masked. However, a recording is a report of an acoustic event, and some of the elements or features of the original sounds must have been available to contemporary listeners, or they would not have accepted the recordings in the first place. Similarly, we may want to retrieve these elements, and for that purpose it is important to identify them. The paper points out some of the fundamental elements of the sounds that have been recorded and discusses the degree to which they may be retrieved in modern replay, either directly from original recordings or via signal-processed transfers. Reference is made to work by D. C. Miller based on his early recognition that the recording process transforms the elements it is desired to retrieve. In a similar manner that Miller compensated his recordings for measurement purposes, we may today compensate early recordings for replay purposes. The presentation will be accompanied by very short comparative excerpts of historical sound recordings.

1:20

2pMU2. An overview of statistical-model-based techniques for audio restoration. Patrick J. Wolfe (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, UK)

This presentation will describe the state of the art in model-based approaches to audio restoration, with an emphasis on results and an explanation of theory suitable for the nonspecialist. An overview of work conducted over the past several years by members of the Cambridge Signal Processing Group will be presented, including techniques for the detection, correction, and interpolation of degraded or missing audio data. The latest advances in a fully Bayesian statistical methodology for modeling of music and speech signals will also be detailed, including a common unifying framework for the treatment of both global and localized audio degradations.

1:40

2pMU3. A two-stage approach to removing noise from recorded music. Jonathan Berger, Maxim J. Goldberg, Ronald C. Coifman (CCRMA, Stanford Univ., Stanford, CA 94305, brg@ccrma.stanford.edu), Maxim J. Goldberg (Ramapo College of New Jersey, Mahwah, NJ 07430), and Ronald C. Coifman (Yale Univ., New Haven, CT 06520)

A two-stage algorithm for removing noise from recorded music signals (first proposed in Berger *et al.*, ICMC, 1995) is described and updated. The first stage selects the “best” local trigonometric basis for the signal and models noise as the part having high entropy [see Berger *et al.*, J. Audio Eng. Soc. **42**(10), 808–818 (1994)]. In the second stage, the original source and the model of the noise

obtained from the first stage are expanded into dyadic trees of smooth local sine bases. The best basis for the source signal is extracted using a relative entropy function (the Kullback–Leibler distance) to compare the sum of the costs of the children nodes to the cost of their parent node; energies of the noise in corresponding nodes of the model noise tree are used as weights. The talk will include audio examples of various stages of the method and proposals for further research.

2:00

2pMU4. Reconstruction of mechanically recorded sound by image processing. Vitaliy Fedeyev, Carl Haber, Zachary Radding (Lawrence Berkeley Natl. Lab., Berkeley, CA 94270), Christian Maul (TaiCaan Technologies Ltd., Southampton, UK), John McBride (Univ. of Southampton, Southampton, UK), and Mitchell Golden

Audio information stored in the undulations of grooves in a medium such as a phonograph record or cylinder may be reconstructed, without contact, by measuring the groove shape using precision optical metrology methods and digital image processing. In this approach, audio signal processing is accomplished by two- or three-dimensional image analysis and processing. The viability of these methods was recently demonstrated on a 78 rpm shellac disc using two-dimensional image acquisition and analysis methods [V. Fedeyev and C. Haber, *J. Audio Eng. Soc.* **51**(12), 1172–1185 (2003)]. The present work expands on these results. A three-dimensional reconstruction of mechanically recorded sound is reported. The source material, an Edison cylinder, was scanned using confocal microscopy and resulted in a faithful playback of the recorded information. Methods to accelerate the scan rates and make these techniques practical for use in working archives are reported as well. [Work supported by the Laboratory Technology Research Program (SC-32), within the Office of Science, U.S. Department of Energy under Contract No. DE-AC03-76SF00098.]

2:20

2pMU5. Source identification and manipulation in stereo music recordings using frequency-domain signal processing. Carlos Avendano (Creative Adv. Technol. Ctr., Scotts Valley, CA 95066)

A short-time frequency domain framework for source identification, separation, and manipulation in stereo music recordings is presented. Using a simplified model of the stereo mix, a similarity measure between the short-time fourier transforms (STFTs) of the input signals is computed to identify time–frequency regions occupied by each source based on the panning coefficients assigned to it during the mix. Individual sources are identified and manipulated by clustering time-frequency components with a given panning coefficient and frequency range. After modification, an inverse STFT is used to synthesize a time-domain processed signal. Applications of the technique to source suppression, enhancement and repanning will be described, and audio demonstrations will be presented to illustrate the results.

2:40

2pMU6. SmartMusicKIOSK: Music-playback interface based on chorus-section detection method. Masataka Goto (Natl. Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-1-1 Umezono, Tsukuba, Ibaraki 305-8568, Japan, m.goto@aist.go.jp)

This paper describes *SmartMusicKIOSK*, a new music-playback interface for trial listening. Traditionally in music stores, customers often search out the chorus or “hook” of a song by repeatedly pressing the fast-forward button, rather than passively listening to the music. This activity is not well-supported by current technology. This research achieves a function for jumping to the chorus section and other key parts of a song, plus a function for visualizing song structure. These functions eliminate the hassle of searching for the chorus and make it easier for a listener to find desired parts of a song, thereby facilitating an active listening experience. This interface, which enables a listener to look for a section of interest by interactively changing the playback position, is useful not only for trial listening but also for more general purposes in selecting and using music. The proposed functions are achieved through an automatic audio-based chorus-section detection method that can detect all the chorus sections by analyzing relationships between various repeated sections in a song. It can also detect modulated chorus sections by introducing an acoustic similarity that enables modulated repetition to be judged correctly. The results of implementing this method in *SmartMusicKIOSK* have demonstrated its usefulness.

3:00

2pMU7. Music processing above and below the fundamental frequency. Marios Athineos and Daniel P. W. Ellis (LabROSA, Dept. of Elec. Eng., Columbia Univ., New York, NY 10027)

Because of the centrality of pitch to music, most music signal processing has been focused on pitch-related timescales, e.g., short-time analysis over 30–50-ms windows. But there is, of course, important information at other scales as well. By looking at ways to model and extract very fine time-scale information one can extract attributes that lead to the perception of “texture” in unpitched instruments such as maracas. At the other extreme, interesting effects can be revealed by investigating the effect of “modulation-domain” processing, such as filtering subband energy envelopes in the 0.1–10-Hz range. A single representation, frequency-domain linear prediction, gives convenient access to both these scales: its use in music analysis/synthesis will be illustrated by examples.

Session 2pNSa**Noise and Psychological and Physiological Acoustics: Noise Impact Evaluation: Old and New II**

Brigitte Schulte-Fortkamp, Cochair

Institute of Technical Acoustics, Technical University Berlin, Secr TA 7, Einsteinufer 25, 10587 Berlin, Germany

Klaus Genuit, Cochair

*HEAD Acoustics GmbH, Eberstrasse 30a, Herzogenrath 52134, Germany***Chair's Introduction—1:15*****Invited Papers*****1:20****2pNSa1. Meaningless artificial sound and its application in urban soundscape research.** Bert De Coensel and Dick Botteldooren (Acoust. Group, Dept. of Information Technol., Ghent Univ., Belgium, bert.decoensel@intec.ugent.be)

Urban areas are increasingly being overwhelmed with uninteresting (traffic) noise. Designing a more matching soundscape for urban parks, quiet backyards, shopping areas, etc., clearly deserves more attention. Urban planners, being architects rather than musical composers, like to have a set of “objective” indicators of the urban soundscape at their disposal. In deriving such indicators, one can assume that the soundscape is appreciated as a conglomerate of sound events, recognized as originating from individual sources by people evaluating it. A more recent line of research assumes that the soundscape as a whole evokes particular emotions. In this research project we follow the latter, more holistic view. Given this choice, the challenge is to create a test setup where subjects are not tempted to react to a sound in a cognitive way, analyzing it to its individual components. Meaningless sound is therefore preferred. After selection of appealing sounds for a given context by subjects, objective indicators can then be extracted. To generate long, complex, but meaningless sound fragments not containing repetition, based on a limited number of parameters, swarm technology is used. This technique has previously been used for creating artificial music and has proved to be very useful.

1:40**2pNSa2. Subjective soundscapes qualitative research in the experience and evaluation of environmental noise.** Uwe Flick (Alice Salomon Univ. of Appl. Sci., Alice-Salomon-Platz 5D-12627, Berlin, Germany, Uwe.Flick@Tu-Berlin.de)

If the subjective experience and evaluation of environmental noise shall be considered and integrated into the current soundscape research, the use of qualitative research methods used in sociology and psychology will become necessary. A triangulation of research methods for measuring objective noise and for the subjective evaluation of noises and sounds on the background of subjective meanings of health and healthy living will be a fruitful way to a more comprehensive understanding of the phenomenon of soundscapes in the context of health and quality of life. In this contribution, a selection of qualitative research methods will be presented that allows for analyzing subjective experiences with environmental noise. Interviews focusing on narratives of episodes and situations (e.g., the episodic interview, Flick, 2002) will be outlined. Issues of how to assess the quality of qualitative research and its results will be addressed and finally the benefits and limits of the triangulation of different methods (e.g., interviews and focus groups or interviews and physical measures) will be discussed. Research experiences from the author's recent studies on health concepts of health professionals will be used for illustration.

2:00**2pNSa3. A cognitive approach to soundscape research.** Daniele Dubois, Catherine Guastavino, Valerie Maffiolo, Manon Raimbault (CNRS LCPE/LAM 11 rue de Lourmel (F) 75015 Paris, France, ddubois@ccr.jussieu.fr), Catherine Guastavino (McGill Univ., Canada), Valerie Maffiolo (FTR&T Lannion, France), and Manon Raimbault (INRETS, Lyon, France)

The present research on cognitive categories for soundscapes focuses on their interpretations and can be seen as mediating between individual sensory experiences and collective representations shared in language and elaborated as knowledge. Results of field inquiries in Paris, Lyon, and Nantes are presented together with results from categorization of recorded soundscapes in laboratory conditions. Categories were identified by means of linguistic analyses of verbal comments and mathematical analyses of similarity judgments. Results indicate that people categorize environmental sounds on the basis of semantic features, namely source identity and pleasantness judgments, rather than perceptual features. Effects of noise on human subjectivity cannot be quantitatively measured thoroughly in terms of physical parameters: auditory judgments depend upon the meaning attributed to acoustic phenomena and noise sources, rather than on inherent properties of the acoustic signal. These findings highlight the fact that an acoustic phenomenon can be diversely conceptualized and lexicalized as cognitive representations. Finally, methodological and theoretical consequences of these findings are established as the basis for further research on soundscape, in order to account not only for noise annoyance but also for sound quality of urban life.

2pNSa4. Soundscapes and their influence on inhabitants—New findings with the help of a grounded theory approach. Andre Fiebig and Brigitte Schulte-Fortkamp (Inst. of Tech. Acoust., TU-Berlin, Einsteinufer 25, D-10587 Berlin, Germany, a.fiebig@gmx.de)

Recent research work has shown the relevance of soundscapes with respect to community noise. A study was conducted aiming at clarifying and improving the comprehension between the acoustical stimulus and assessment, response, attitude, and perception of inhabitants in urban environments regarding their urban soundscapes. Narrative and issue-centered interviews were carried out to identify common perceptive patterns. Therefore subjects were required to articulate their opinions and sensations about their surroundings including environmental sound. At the same time acoustic measurements were performed. Therefore the study employs methodology of qualitative-interpretative research, since perception is constantly embedded in social contexts. By means of the “grounded theory” approach, a general methodology of qualitative social analysis, individual and interindividual determinants have been explored in the process of evaluating urban soundscapes based on a new integrative diagram. The new model will be presented.

2pNSa5. The acoustical diary as an innovative tool in soundscape evaluation. Brigitte Schulte-Fortkamp (Inst. of Tech. Acoust., TU-Berlin, Einsteinufer 25, D-10587 Berlin, Germany, brigitte.schulte-fortkamp@tu-berlin.de) and Klaus Genuit (HEAD acoustics GmbH, D-52120 Herzogenrath, Germany)

A new field study evaluating soundscapes investigates closely the reactions of traffic noise with a particular regard to the street surface. The combination of methods with different sensibilities for the subject’s process of perceiving and evaluating noise in such ambiances is necessary for a reliable and valid analysis and interpretation of data. Acoustic measurements are carried out in critical segments of the street as well as in the respective apartments of the inhabitants, which are questioned in narrative interviews. The acoustic measurements are taken simultaneously in the apartment and on the street. Apartments were selected which issue into the street; outside measurements are performed in front of the buildings on the sidewalk. During the interviews in the apartments the occurring noises are registered by noisebook. As a rule the measurement spot within the apartment is the area in which the interviewed person mostly resides, when he/she takes repose. Further analysis points out the importance of an extended evaluation with an acoustical diary which combines technical and sociological measurement procedures. Performance of the entire data collection process and first results will be discussed.

Contributed Papers

2pNSa6. House attenuation of aircraft and street traffic noise in relation to annoyance: A pilot study. Karl D. Kryter (Dept. of Commun. Disord., San Diego State Univ., San Diego, CA 92182, kdkryter@earthlink.net)

House barrier sound attenuation was measured as differences between the levels of street traffic and aircraft noise found outdoors near the front (street facing) and rear façades, and in rooms of two one-story houses. Also, attenuation of street traffic and aircraft noise between the front and rear façades of similar one-story houses was calculated in accordance with acoustical models. Findings: (1) relative to levels measured near the front façade, street traffic noise near the rear façade, and in rooms inside the houses, was, on average, 9 max dB and 8 SEL dB more attenuated than was aircraft noise. (2) When attenuation of street traffic noise was measured relative to its level near the front façade, and attenuation for aircraft noise was measured relative to its level near the rear façade, street traffic noise was, on average, 7 max dB and 6 SEL dB more attenuated than aircraft noise. (3) Calculated barrier attenuation of street traffic noise was, on average, 1.3 max dB less than measured; calculated barrier attenuation of aircraft noise differed from measured by ~0.4 max dB. Implications of the findings for guidelines for assessing impact of these noises on communities and land-use zoning are discussed.

2pNSa7. The effect of spatial distribution on the annoyance caused by simultaneous sounds. Joos Vos, Adelbert W. Bronkhorst (Dept. of Percept. TNO Human Factors, P.O. Box 23, 3769 ZG Soesterberg, The Netherlands), and Thomas Fedtke (Physikalisch-Technische Bundesanstalt, Bundesallee 100, D-38116 Braunschweig, Germany)

A considerable part of the population is exposed to simultaneous and/or successive environmental sounds from different sources. In many cases, these sources are different with respect to their locations also. In a laboratory study, it was investigated whether the annoyance caused by the multiple sounds is affected by the spatial distribution of the sources. There were four independent variables: (1) sound category (stationary or moving), (2) sound type (stationary: lawn-mower, leaf-blower, and chain saw; moving: road traffic, railway, and motorbike), (3) spatial location (left, right, and combinations), and (4) A-weighted sound exposure level (ASEL of single sources equal to 50, 60, or 70 dB). In addition to the individual sounds in isolation, various combinations of two or three different sources within each sound category and sound level were presented for rating. The annoyance was mainly determined by sound level and sound source type. In most cases there were neither significant main effects of spatial distribution nor significant interaction effects between spatial distribution and the other variables. It was concluded that for rating the spatially distrib-

uted sounds investigated, the noise dose can simply be determined by a summation of the levels for the left and right channels. [Work supported by CEU.]

3:45

2pNSa8. Evaluation of urban soundscape by future architects. Mei Zhang and Jian Kang (School of Architecture, Sheffield Univ., Western Bank, Sheffield S10 2TN, UK, m.zhang@sheffield.ac.uk)

Urban soundscape design has drawn great attention along with the ever increasing urban noise level. Previous research shows that people with different social and demographic backgrounds may have different sound preferences in urban environment. For example, gender, age, and cultural background have been proved to affect peoples soundscape evaluation. In this research the soundscape evaluation by a group of architectural students was made, investigating how the future architectural/urban designers value the urban soundscape they experience everyday, and how they would like to design urban soundscape. The soundscape evaluation/walk was conducted with 60 students in a number of urban open public spaces in Sheffield, UK. The evaluation was both for the overall soundscape and for individual sounds. Eighteen indices with seven-point bipolar rating scale were used. Both connotative meanings of urban environment sounds, such as calming-agitating, interesting-boring and like-dislike, and denotative meanings, such as quiet-noisy, sharp-flat and smooth-rough, were considered. In addition, the students were asked to give design suggestions. The semantic differential technique was applied for the analysis.

Comparison was also made between this special group and general public. The results are useful for studying the links/interactions between general architectural/urban design and the soundscape design. [Work supported by British Academy.]

4:00

2pNSa9. Evaluation of the performance of the blast analysis and measurement system. George A. Luz (U.S. Army Ctr. for Health Promotion and Preventive Medicine, Aberdeen Proving Ground, MD 21010-5403, George.Luz@amedd.army.mil)

In the years since the introduction of the C-weighted day-night average sound level (DNL) to assess the noise of military explosives, Army practice has evolved to incorporate linear peak sound-pressure level into the evaluation of military training noise. Although the DNL remains as the method of choice for National Environmental Policy Act (NEPA) documentation and for land-use planning, peak level is used by firing range operators for day-to-day complaint management. Several different monitoring system designs are being used at Army installations to provide range operators with real-time feedback on blast noise levels in nearby residential areas. One of these, the Blast Analysis and Measurement (BLAM) system, is a modified version of a sonic boom monitor designed by the U.S. Air Force. Data collected from two BLAM units located near a 120-mm tank gunnery range were evaluated in terms of hit rate and false-alarm rate over a range of 94 to 140 decibels linear peak. Hit- and false-alarm rates are compared with hit- and false-alarm rates reported for other blast noise monitoring system designs.

TUESDAY AFTERNOON, 25 MAY 2004

LIBERTY 1/2, 3:25 TO 5:00 P.M.

Session 2pNSb

Noise: Topics in Active Noise Control

Scott D. Sommerfeldt, Chair

Department of Physics and Astronomy, Brigham Young University, Provo, Utah 84602-4673

Chair's Introduction—3:25

Contributed Papers

3:30

2pNSb1. A linear independence method for system identification/secondary path modeling for active control. Benjamin Kim and David Swanson (Appl. Res. Lab. and Grad. Prog. in Acoust., The Penn. State Univ., University Park, PA 16802, benkim@psu.edu)

A novel method for noninvasive system identification/secondary path modeling has been developed for single- and multi-channel filtered-x LMS-based active noise control (ANC). The problem of on-line secondary path modeling is recognized as one of linear dependence associated with an underdetermined system, a one-equation/two-unknown problem in which the highly correlated primary source and secondary source contributions to the error signal are not readily distinguishable. The method resolves this uniqueness issue by introducing a second equation with similar unknowns. The critical linear independence of the two equations, hence the proposed designation, is achieved with a single perturbation of the control filter output thereby making the system solvable. This new secondary path modeling strategy was implemented using a novel real-time DSP control architecture and tested on a novel transducer-less system devised to investigate the behavior of ANC algorithms. Results of narrow-band, broadband, and multi-channel tests reveal estimates of exceptional accuracy, in both magnitude and phase; any bias in the secondary path esti-

mate, whether introduced by the primary noise source or other secondary sources in the system, is eliminated using this method. Its quickness as well contributes significantly to the stability and performance of filtered-x LMS-based controllers. [Work supported by ONR.]

3:45

2pNSb2. Hybrid feedforward-feedback active noise control for circumaural headsets. Alexander D. Streeter, Laura R. Ray, and Robert D. Collier (Thayer School of Eng., Dartmouth College, Hanover, NH 03755)

Traditional stability-performance tradeoffs pose limitations on active noise reduction (ANR) using feedback control, which are evident in circumaural communication headsets. In these systems, the cavity resonant behavior necessitates low feedback gains, reducing performance. Feedforward ANR using a Lyapunov-tuned least-mean-square filter dramatically enhances noise reduction performance compared with feedback ANR [Cartes *et al.*, *J. Acoust. Soc. Am.* **111**, 1758–1771 (2002); Collier *et al.*, *NOISECON* (2003)]. However, feedforward performance is sensitive to the noise source stationarity, and the frequency-dependent forward path gain reduces stability margins. This paper presents experimental results for a hybrid feedforward-feedback ANR system, which enhances performance

and gain margins for both stationary and nonstationary noise. Algorithms are optimized and measurements are performed with Thayer's rapid prototyping system and associated low-frequency acoustic test cell using a circumaural hearing protector. In the frequency range 50–800 Hz, the hybrid system provides an average of 27 dB active noise reduction (40 dB total) for tonal noise and 17 dB reduction (32 dB total) for nonstationary noise. Performance below 100 Hz improves by as much as 15 dB over that of individual control components, and gain margin of the hybrid system improves substantially over individual feedforward or feedback components.

4:00

2pNSb3. Active noise control of small axial cooling fans. Brian B. Monson, Scott D. Sommerfeldt, and Connor Duke (Brigham Young Univ., N283 ESC, Provo, UT 84602, bbm9@email.byu.edu)

An active noise control system has been developed for the reduction of noise radiated by small axial cooling fans, such as those found in desktop computers. This system, based on a modified version of the filtered-x LMS algorithm, uses four small actuators surrounding the fan in a mock computer casing. Global attenuation of the fan noise has been demonstrated using four-channel control. Due to industry volume constraints, a smaller system is required to replace the existing one, while still maintaining similar performance. The system is designed with a smaller fan with small actuators, so as not to exceed the size of the original fan. In order to maintain comparable air flow, the smaller fan must run at higher speeds, causing potentially greater output noise levels at higher frequencies. Global attenuation of these levels is desired. Implementation and experimental results of the system will be presented. A comparison of the two systems, and the feasibility of the replacement of the larger fan by the active control system with a smaller fan will be discussed.

4:15

2pNSb4. Aircraft interior ANC with light-weight actuators. Christian Gerner, Delf Sachau (Helmut-Schmidt-Univ., Mechatronics, Holstenhofweg 85, D-22043 Hamburg, Germany, gerner@unibw-hamburg.de), and Harald Breitbach (Airbus Germany GmbH, D-21229 Hamburg, Germany)

In propeller-driven aircraft the main source for internal noise is tonal disturbances caused by the propeller blades that are passing the fuselage. In a certain four-propeller military transport aircraft the maximum sound level in the cabin can reach up to 110 dB(A), not taking into account any noise control treatments. Inside the semiclosed loadmaster working station (LMWS) the sound level must be reduced down to 86 dB(A). It is proposed to reach this goal with an active noise control system, because passive solutions are too heavy at low frequencies. The optimal positions of the loudspeakers are found by finite-element calculations. These positions have been realized in a full-scale test bed. In the test bed a reduction of the sound-pressure level of more than 30 dB within a specified volume was achieved at a frequency of 100 Hz. In this test bed hi-fi speakers are

used as secondary actuators. These speakers are heavy and of unsuitable geometric dimensions for an aircraft. Therefore, other actuators, e.g., flat panel speakers, will be investigated with respect to the application to a mock-up of the LMWS.

4:30

2pNSb5. Limitations on the performance of active noise control systems due to subjective effects. Gerard Mangiante (Laboratoire de Mécanique et d'Acoustique, CNRS, 31 Chemin Joseph-Aiguier, 13402 Marseille Cedex 20, France, mangiante@lma.cnrs-mrs.fr)

Although numerous descriptions of the physical effects of active noise control can be found in the literature, there are few papers available about the subjective effects of this control on listeners. Due to these effects, a significant reduction in the sound pressure level of a primary noise may result in a negligible decrease in the signal perceived by a human observer. This paper reports on the performance of an active system, expressed in terms of reduction in loudness level. For that purpose, the equivalent loudness levels before and after active control were computed for various test signals. For pure tones, the calculation of the loudness level was based on Robinson and Dadson equal-loudness contours; for complex signals, this calculation used the ISO Method A. It was shown that the reduction in loudness level due to active control is highly dependent of the sound pressure level of the primary noise: For frequencies lower than 1000 Hz, the subjective reduction decreases with increasing primary noise level. The subjective reduction was also calculated as a function of the difference in amplitude and phase between the primary and secondary pressure variations. The reduction expressed in phons was then compared to that expressed in dB.

4:45

2pNSb6. Acoustic feedback cancellation for active noise control in pipe. Yun-Hui Liu and Chuan-Yu Hung (Southern Taiwan Univ. of Technol., 1 Nan-Tai St., Yung-Kang City, Tainan Hsien, Taiwan, ROC 710, yhliu@mail.stut.edu.tw)

In most practical applications of active noise control, acoustic feedback is a major problem that often interferes with the operation of the control system and even renders it unstable. The optimal collocated positions of secondary source and two microphones are studied and the theoretical equations that reduce the influence of acoustic feedback are developed based on plane-wave transmission theory of sound in a duct. The original signal of primary noise is obtained from the measured signals of reference microphone and error microphone, which are transformed to frequency domain by FFT and operated. In this study, the performance of the proposed method is compared with other traditional time-domain methods by means of simulation and analysis using LABVIEW programming language. The results showed that the proposed method can effectively reduce the influence of acoustic feedback. In practical application of active noise control in a duct, the adaptive control or other control method incorporating the proposed method in this paper will improve the stability and performance effectively.

Session 2pPP

Psychological and Physiological Acoustics: Auditory Periphery and Models

Laurel H. Carney, Chair

Institute for Sensory Research, Syracuse University, 621 Skytop Road, Syracuse, New York 13244-5290

Contributed Papers

1:30

2pPP1. Coupling of earphones to human ear and to coupler. Dejan Ciric (Faculty of Electron. Eng., Beogradska 14, 18000 Nis, Serbia and Montenegro, dciric@elfak.ni.ac.yu) and Dorte Hammershoi (Aalborg Univ., DK-9220 Aalborg O, Denmark)

The use of a standardized acoustical coupler should enable a calibration of audiometric earphones which ensures that the thresholds determined in the audiometry will be independent of the earphone type. This requires that the coupler approximates the average human ear closely. Nevertheless, the differences among earphones as well as between human ears and the coupler affect the results of audiometric measurements inducing uncertainty. The influence of these differences is examined by investigating the coupling of different earphones to human ears and to the standardized coupler. This is done by measurement of the transfer functions from input voltage of the earphone terminals to the entrance of the ear canal in two situations: (1) open, and (2) blocked. Similar measurements were carried out with the coupler, but since the "ear-canal entrance" is not well-defined for the coupler, the mentioned measurements were done at different depths in the coupler. The earphone's coupling to (i) human ears and to (ii) the coupler, described in terms of the pressure division at the entrance of the ear canal, were compared. The results indicate that the coupling to the human ear and the coupling to the standardized coupler differ.

1:45

2pPP2. Using reflectance phase to estimate the acoustic response at the tympanic membrane. Douglas H. Keefe (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, keefe@boystown.org)

A common problem in ear-canal acoustics is to estimate the SPL and acoustic transfer functions (ATFs) at the tympanic membrane (TM) based on a response measured at a distal probe location, but standing waves complicate interpretation above 1.5 kHz. Transmission is well approximated up to 8 kHz by modeling the ear canal as a rigid-walled cylindrical tube; the model transforms measurements at the probe tip to those at the TM. Based on SPL and ATFs measured at the probe in 42 normal-hearing and 18 conductive-impaired ears from 0.25–8 kHz, the equivalent SPL and ATFs at the TM are calculated for ambient and pressurized conditions in the ear canal. Tube area is calculated based on probe-tip size, while tube length from probe to TM is calculated from the group delay of the reflectance phase averaged from 6 to 8 kHz. In normal-hearing ears, the mean (\pm standard deviation) of the SPL difference at the TM *re*: probe has a maximum of 9.2 ± 3.8 dB at 3.7 ± 0.8 kHz. The equivalent TM admittance has a wideband shape similar to *in vivo* measurements of umbo velocity level at constant SPL at the TM [Ruggero and Temchin, 2003]. [Work supported by NIH.]

2:00

2pPP3. The impact of cochlear fine structure on hearing thresholds and DPOAE levels. Jungmee Lee, Glenis Long (Speech and Hearing Sci., Grad. Ctr. of CUNY, 365 5th Ave., New York, NY 10016, jmlee6@msn.com), and Carrick L. Talmadge (The Univ. of Mississippi, University, MS 38677)

Although otoacoustic emissions (OAE) are used as clinical and research tools, the correlation between OAE behavioral estimates of hearing status is not large. In normal-hearing individuals, the level of OAEs can vary as much as 30 dB when the frequency is changed less than 5%. These pseudoperiodic variations of OAE level with frequency are known as fine structure. Hearing thresholds measured with high-frequency resolution reveals a similar (up to 15 dB) fine structure. We examine the impact of OAE and threshold fine structures on the prediction of auditory thresholds from OAE levels. Distortion product otoacoustic emissions (DPOAEs) were measured with sweeping primary tones. Psychoacoustic detection thresholds were measured using pure tones, sweep tones, FM tones, and narrow-band noise. Sweep DPOAE and narrow-band threshold estimates provide estimates that are less influenced by cochlear fine structure and should lead to a higher correlation between OAE levels and psychoacoustic thresholds. [Research supported by PSC CUNY, NIDCD, National Institute on Disability and Rehabilitation Research in U.S. Department of Education, and The Ministry of Education in Korea.]

2:15

2pPP4. Can tone-burst otoacoustic emissions be used to measure basilar-membrane response in humans? Michael Epstein, Søren Buus (ECE Dept. (440 DA), Inst. of Hearing, Speech and Lang. and Commun. and DSP Ctr., Northeastern Univ., Boston, MA 02115, mepstein@ece.neu.edu), and Mary Florentine (Northeastern Univ., Boston, MA 02115)

Tone-burst otoacoustic emissions (TBOAEs) are a potential tool for examining basilar-membrane activity in humans. However, their use requires knowledge of just how the relation between TBOAEs and basilar-membrane input-output (BM I/O) activity depends on measurement and analysis parameters. The present experiment examined three such parameters: (1) response-window timing, (2) response delinearization, and (3) stimulus frequency. Varying the onset of the post-stimulus-recording analysis window had little effect on the function relating emission amplitude to SPL. Response delinearization, if anything, tended to obscure the relation between TBOAE and stimulus level. Changes in stimulus frequency over a range of 1 Bark had no effect on TBOAE amplitude. Although none of these experimental manipulations had a significant effect on TBOAE measurements, a judicious choice of the experimental parameters can optimize the assessment of BM I/O functions. It is concluded that TBOAEs provide a useful tool for objective BM I/O function analysis in humans. [Work supported by NIH-NIDCD Grant R01DC00187.]

2pPP5. Relating psychophysical and otoacoustic emission suppression tuning curves. Jeffrey J. DiGiovanni and Erika M. Zettner (School of Hearing Speech and Lang. Sci., Ohio Univ., Athens, OH 45701, digiovan@ohio.edu)

Psychophysical tuning curves (PTCs) are a well established method for estimating frequency selectivity in humans. PTCs are closely related to the shape of neural tuning curves. Furthermore, these two measurements are believed to represent frequency resolution processes primarily occurring in the cochlea. More recently, otoacoustic emission suppression tuning curves (STCs) have been measured and found to be characteristically similar in shape and tip frequency to PTCs (Abdala *et al.*, 1996). STCs may represent an objective, noninvasive, measure of initial stages of cochlear tuning. The goal of this study was to compare and quantify the difference between transient evoked STCs (TE-STC) and PTCs. TE-STCs and PTCs were obtained in each normal-hearing subject with 4000-Hz probe tones. Preliminary data show that Q10 values and tip levels follow the same trend for both tuning curves. However, actual values of these parameters differ. Implications of methodological differences between these two paradigms will be discussed.

2pPP6. On the possible role of MOC efferents in speech reception in noise. Oded Ghitza (Sensimetrics Corp., 48 Grove St., Somerville, MA 02144, oded@sens.com)

The hypothesis explored in this study is that the MOC efferent system plays an important role in speech reception in the presence of sustained background noise. This talk describes efforts to assess this hypothesis using a test of initial consonant reception (the Diagnostic Rhyme Test) performed by subjects with normal hearing. Activation of selected parts of the efferent system was attempted by presenting speech and noise in various configurations (gated/continuous, monaural/binaural). Initial results of these experiments show a gated/continuous difference analogous to the "masking overshoot" in tone detection. These results are interpreted to support the hypothesis of a significant efferent contribution to initial phone discrimination in noise. [Work supported by AFOSR.]

2pPP7. Revisiting the Fano factor in peripheral neurons. Willy Wong (Dept. of Elec. and Computer Eng., and Inst. of Biomaterials and Biomed. Eng., Univ. of Toronto, ON M5S 3G4, Canada, willy@eecg.utoronto.ca)

Despite the wealth of studies on the dynamic characteristics of peripheral auditory neurons, very little has been reported on the higher statistical moments of the neural spike train. The notable exception is the study by Teich and Khanna (1985) where both the mean and the variance of the neural count are reported. The simplest model one can ascribe to a neural spike train is a homogeneous Poisson process. However, experimental data do not bear out such predictions. Other models have been proposed but the general consensus is that the underlying process is far from simple. We offer an alternative account of the fluctuations that occur at the peripheral level. Our explanation does not rely on assumptions regarding the process underlying individual spikes. Instead, we make use of the information-theoretical model of the neuron that we have been developing over the past 10 years (Norwich and Wong, 1995; Wong, 1997). The two key results predicted by the model are that (a) the mean-variance ratio has an approximate value of 2 and (b) the distribution governing the neural count is Gaussian to a good approximation.

2pPP8. Towards an analytical derivation of a computational auditory masking model. Joerg M. Buchholz (Univ. of Western Sydney, MARCS Auditory Labs., Locked Bag 1797, Penrith South DC NSW 1797, Australia, j.buchholz@uws.edu.au)

Due to the fact that the signal processing underlying auditory masking is highly nonlinear, existing auditory masking models are usually designed on rather a trial-and-error basis than in a well-defined analytical way. Employing a model structure, which is based on the concept of signal-dependent compression (SDC), a method is presented which demonstrates the analytical derivation of a computational auditory masking model. Given the SDC-based model approach, analytical approximations of the model's masked threshold simulations can be derived, which can directly be compared to masking functions known from the relevant literature. In this way, the present model structure has been adjusted in order to describe mainly two masking functions: (i) Webber's law, which approximates the masker level dependency of simultaneous noise-on-tone masking, and (ii) a function proposed by Jestead *et al.*, *J. Acoust. Soc. Am.* **71**, 950-962 (1982), which approximates the masker level and test-signal delay dependency of forward masking. The practical applicability of the derived model structure has been verified by comparing model simulations to various known psychoacoustical data of simultaneous masking and forward masking for broadband noise masking tones.

2pPP9. Modeling the detection and level discrimination of tones in the presence of masking noise. Xuedong Zhang (Dept. of Biomed. Eng. and Hearing Res. Ctr., Boston Univ., Boston, MA 02215, zxd@bu.edu) and Laurel H. Carney (Syracuse Univ., Syracuse, NY)

The performance of three model detectors was evaluated for masked detection and level discrimination of tones in noise with different bandwidths and in a roving level paradigm (Kidd *et al.*, 1989). The model detectors were based on (1) the output energy of several auditory filters tuned to different frequencies, (2) the envelope statistics (peakiness) of the auditory filter outputs, and (3) the cross-correlations between auditory-nerve (AN) fibers in a population model. The energy-based model predicted detection thresholds, but failed to predict masked level discrimination; this detector was robust for roving-levels in the wideband condition by combining information from different auditory filter outputs and the prediction was affected by roving narrow-band masker levels. Thresholds of the envelope-based detector were worse than human thresholds for some conditions; however, this model was robust in roving-levels for both wideband and narrow-band maskers. The monaural cross-correlation detector included model cells that were sensitive to temporal cues in model AN inputs in response to wideband noise and also included cells that were sensitive to input level changes. This model naturally combined rate and temporal information and predicted performance for both masked detection and discrimination. [Work supported by NIDCD R01-01641.]

2pPP10. Physiologic correlates to background noise acceptance. Joanna Tampas, Ashley Harkrider, and Anna Nabelek (Dept. of Audiol. and Speech Pathol., Univ. of Tennessee, 457 S. Stadium Hall, Knoxville, TN 37996)

Acceptance of background noise can be evaluated by having listeners indicate the highest background noise level (BNL) they are willing to accept while following the words of a story presented at their most comfortable listening level (MCL). The difference between the selected MCL and BNL is termed the acceptable noise level (ANL). One of the consistent findings in previous studies of ANL is large intersubject variability in acceptance of background noise. This variability is not related to age, gender, hearing sensitivity, personality, type of background noise, or speech perception in noise performance. The purpose of the current experiment was to determine if individual differences in physiological activ-

ity measured from the peripheral and central auditory systems of young female adults with normal hearing can account for the variability observed in ANL. Correlations between ANL and various physiological responses, including spontaneous, click-evoked, and distortion-product otoacoustic

emissions, auditory brainstem and middle latency evoked potentials, and electroencephalography will be presented. Results may increase understanding of the regions of the auditory system that contribute to individual noise acceptance.

TUESDAY AFTERNOON, 25 MAY 2004

LIBERTY 1/2, 1:30 TO 3:10 P.M.

Session 2pSA

Structural Acoustics and Vibration and Noise: Urban Transit Noise

Paul L. Burge, Cochair

Acentech, 33 Moulton Street, Cambridge, Massachusetts 02138

Daniel R. Raichel, Cochair

2727 Moore Lane, Fort Collins, Colorado 80526

Chair's Introduction—1:30

Invited Papers

1:35

2pSA1. New developments in transit noise and vibration criteria. Carl E. Hanson (Harris Miller Miller & Hanson, Inc., 15 New England Executive Park, Burlington, MA 01803, chanson@hmmh.com)

Federal Transit Administration (FTA) noise and vibration impact criteria were developed in the early 1990's. Noise criteria are ambient-based, developed from the Schultz curve and fundamental research performed by the U.S. Environmental Protection Agency in the 1970's. Vibration criteria are single-value rms vibration velocity levels. After 10 years of experience applying the criteria in assessments of new transit projects throughout the United States, FTA is updating its methods. Approach to assessment of new projects in existing high-noise environments will be clarified. Method for assessing noise impacts due to horn blowing at grade crossings will be provided. Vibration criteria will be expanded to include spectral information. This paper summarizes the background of the current criteria, discusses examples where existing methods are lacking, and describes the planned remedies to improve criteria and methods.

1:55

2pSA2. Vibration criteria for transit systems in close proximity to university research activities. Steven Wolf (Parsons Brinckerhoff, 505 S. Main St., Orange, CA 92868)

As some of the newer LRT projects get closer to research facilities the question arises "how do you assess the potential impact of train operations on the activities within these types of facilities?" There are several new LRT projects that have proposed alignments near or under university research facilities. The traditional ground vibration analysis at these locations is no longer valid but requires a more sophisticated approach to identifying both criteria and impact. APTA, ISO, IES, and FTA vibration criteria may not be adequate for the most sensitive activities involving single cell and nano technology research. The use of existing ambient vibration levels is evaluated as a potential criteria. A statistical approach is used to better understand how the train vibration would affect the ambient vibration levels.

2:15

2pSA3. Determination of noise and vibration impacts of construction and operation of the Second Avenue Subway. Jessica L. Pristera (AKRF, Inc., 117 E. 29th St., New York, NY 10016, jessica_pristera@akrf.com)

An acoustical study was conducted to determine the potential for airborne noise and ground-borne noise and vibration impacts generated by construction and operation of the Second Avenue Subway. The study was performed in support of an environmental impact statement (EIS) that defined the areas along the proposed Second Avenue Subway corridor where any significant impacts would occur as a result of construction activity and operation of the Second Avenue Subway. Using FTA guideline procedures, project-generated noise levels from subway construction and operations were determined. Construction noise levels exceeded operational noise levels. With limited alternative construction methods, practical mitigation methods were determined to reduce impacts.

2pSA4. Update on Chicago Transit Authority noise, how the city is attacking the problem. Matthew McDuffee and Dominique Cheenne (Columbia College Chicago, 600 S. Michigan Ave., Chicago, IL 60605)

Chicago's light rail rapid transit system is known as the L. The city of Chicago has the oldest standing tracks in the country with some portions of the system dating back to the 1880s. The noise of the L is a signature mark of Chicago and has long been dealt with by residents due to the integral nature of the transportation system to the city. The paper outlines what the CTA (Chicago Transit Authority) planning and development office defines as the four major sources of noise, and how they deal with the issues. Among these are thump, roar, rumble, and squeal. These issues are dealt with through rail grinding programs, wheel truing, and structural improvements, as well as wheel dampening rings, rail lubrication, and reconstruction projects to reduce the steepness of curves. A before and after study of one of the recent CTA construction projects was done to see if these goals are being met by the reconstruction projects being performed in the field. The CTA has agreed to cooperate with me in this project; the Vice President of planning and development has offered data and the CTA approval of the independent study.

Contributed Paper

2:55

2pSA5. Noise annoyance caused by magnetic levitation train passbys. Joos Vos (Dept. of Percept. TNO Human Factors, P.O. Box 23, 3769 ZG Soesterberg, The Netherlands)

In a laboratory study, the annoyance caused by the passby sounds from a magnetic levitation (maglev) train was investigated. The outdoor A-weighted sound exposure level (ASEL) of the maglev sounds varied from 65 to 90 dB. The driving speed of the maglev train varied from 100 to 400 km/h. Four important results were obtained. Provided that the outdoor ASELs were the same, (1) the annoyance was independent of the

driving speed of the maglev train, (2) the annoyance caused by the maglev train was considerably higher than that caused by intercity trains, (3) the annoyance caused by the maglev train was hardly different from that caused by road traffic (passenger cars and trucks), and (4) the results (1)–(3) held true both for open or closed windows. On the basis of the present results, it might be expected that the sounds are equally annoying if the ASELs of the maglev-train passbys are at least 5 dB lower than those of the intercity train passbys. Consequently, the results of the present experiment do not support application of a railway bonus to the maglev-train sounds. Issues for future research, such as exploring further contributions of nonacoustic factors, will be discussed.

TUESDAY AFTERNOON, 25 MAY 2004

NEW YORK BALLROOM B, 1:00 TO 5:15 P.M.

Session 2pSC

Speech Communication: Speech Perception and Production in Monolingual and Bilingual Acquisition

Linda Polka, Cochair

School of Communication Sciences and Disorders, McGill University, Beatty Hall, 1266 Pine Avenue West, Montreal, Quebec H3G 1A8, Canada

Megha Sundara, Cochair

School of Communication Sciences and Disorders, McGill University, Beatty Hall, 1266 Pine Avenue West, Montreal, Quebec H3G 1A8, Canada

Chair's Introduction—1:00

Invited Papers

1:05

2pSC1. Cross-linguistic experiments in word-form recognition. Marilyn Vihman (Univ. of Wales Bangor, Gwynedd LL57 2AS, Wales, UK, m.vihman@bangor.ac.uk)

When do children first represent word forms without experimental training or contextual support? Both English- and Welsh-learning children were tested, replicating Halle and Boysson-Bardies (1994: French, 11 months). Twelve children acquiring English showed word-form recognition by 11 months (Vihman *et al.*, in press); 12 Welsh children showed the effect at 12 months but a separate sample of 12 tested at 11 months did not (Vihman and DePaolis, 1999). A subsequent study of 16 children using event-related potentials (ERPs) showed word-form recognition within 250 ms for English at 11 months (Thierry *et al.*, 2003). Attempts to locate the age of onset longitudinally proved problematic: Repeated tests of single samples of English and Welsh monolingual children (12 each) at 9, 10, 11, and 12 months showed that infant episodic memory interferes sufficiently with longitudinal observation based on a single set of stimuli to preclude drawing any conclusions. Cross-sectional samples of monolingual English and Welsh children (24 each) are currently being tested at 9 to 12 months, using both head turn and ERPs, as are English/Welsh bilingual children at 11 months. These studies should yield solid information as to the age of onset of spontaneous word form representation. [ESRC support is gratefully acknowledged.]

1:25

2pSC2. Native sound category formation in simultaneous bilingual acquisition. Laura Bosch (Dept. of Psicologia Basica, Univ. of Barcelona, P. Vall d'Hebron, 171, 08035 Barcelona, Spain, laurabosch@ub.edu)

The consequences of early bilingual exposure on the perceptual reorganization processes that occur by the end of the first year of life were analyzed in a series of experiments on the capacity to discriminate vowel and consonant contrasts, comparing monolingual and bilingual infants (Catalan/Spanish) at different age levels. For bilingual infants, the discrimination of target vowel contrasts, which reflect different amount of overlapping and acoustic distance between the two languages of exposure, suggested a U-shaped developmental pattern. A similar trend was observed in the bilingual infants discrimination of a fricative voicing contrast, present in only one of the languages in their environment. The temporary decline in sensitivity found at 8 months for vowel targets and at 12 months for the voicing contrast reveals the specific perceptual processes that bilingual infants develop in order to deal with their complex linguistic input. Data from adult bilingual subjects on a lexical decision task involving these contrasts add to this developmental picture and suggest the existence of a dominant language even in simultaneous bilingual acquisition. [Work supported by JSMF 10001079BMB.]

1:45

2pSC3. Speech perception and production in bilinguals and monolinguals: Convergent and divergent patterns in adulthood and development. Megha Sundara and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., 1266 Pine Ave. W., Montreal, QC H3G 1A8, Canada)

Most investigations of the role of language experience have looked at monolingual subjects. Data are presented from a series of experiments with consonant contrasts to highlight how bilingual adults exposed to Canadian-English and Canadian-French from birth accommodate two sets of input regularities in their perception and production behavior. Results from these investigations indicate that there are similarities as well as differences between adult simultaneous bilinguals and their monolingual peers. Recent findings from our lab exploring phonetic discrimination and word segmentation in monolingual and bilingual infants and children, which provide insights into the developmental roots of these adult patterns, are also presented. The findings from this work illustrate that during the developmental process bilingual infants and children differ from their monolingual peers in rate as well as pattern of development for some perceptual tasks but not others.

2:05

2pSC4. Developmental and communicative factors affecting VOT production in English and Arabic bilingual and monolingual speakers. Ghada Khattab (Speech and Lang. Sci., King George VI Bldg., Newcastle Univ., Queen Victoria Rd., Newcastle upon Tyne NE1 7RU, UK, ghada.khattab@ncl.ac.uk)

VOT patterns were investigated in the production of three Lebanese-English bilinguals' aged 5, 7, and 10, six aged-matched monolingual controls from the bilinguals' immediate communities, and the parents of bilinguals and monolinguals. The aim was to examine the extent to which children exposed to two languages acquire separate VOT patterns for each language and to determine the factors that affect such acquisition. Results showed that VOT patterns for each bilingual child differed significantly across the two languages. But while the contrast in English resembled a monolingual-like model, that for Arabic exhibited persisting developmental features; explanations were offered in terms of the relationship between input and complexity of voicing lead production. Evidence was used from developmental changes that were noted for two of the bilingual subjects over a period of 18 months. English code-switches produced by the bilinguals during Arabic sessions exhibited different VOT patterns from those produced during English sessions, which underlined the importance of taking the language context into consideration. Finally, results from monolinguals and bilinguals showed that the short lag categories for the two languages were different despite a degree of overlap. Such findings require finer divisions of the three universal VOT categories to account for language-specific patterns.

2:25

2pSC5. Acquisition of English word stress patterns in early and late bilinguals. Susan G. Guion (Linguist. Dept., 1290 Univ. of Oregon, Eugene, OR 97403, guion@uoregon.edu)

Given early acquisition of prosodic knowledge as demonstrated by infants' sensitivity to native language accentual patterns, the question of whether learners can acquire new prosodic patterns across the life span arises. Acquisition of English stress by early and late Spanish-English and Korean-English bilinguals was investigated. In a production task, two-syllable nonwords were produced in noun and verb sentence frames. In a perception task, preference for first or last syllable stress on the nonwords was indicated. Also, real words that were phonologically similar to the nonwords were collected. Logistic regression analyses and ANOVAs were conducted to determine the effect of three factors (syllable structure, lexical class, and stress patterns of phonologically similar words) on the production and perception responses. In all three groups, stress patterns of phonologically similar real words predicted stress on nonwords. For the two other factors, early bilinguals patterned similarly to the native-English participants. Late Spanish-English bilinguals demonstrated less learning of stress patterns based on syllabic structure, and late Korean-English bilinguals demonstrated less learning of stress patterns based on lexical class than native-English speakers. Thus, compared to native speakers, late bilinguals' ability to abstract stress patterns is reduced and affected by the first language. [Work supported by NIH.]

2:45-3:00 Break

2p TUE. PM

3:00

2pSC6. Neural network simulation of habituation and dishabituation in infant speech perception. Bruno Gauthier, Rushen Shi, and Robert Proulx (Dept. of Psych., Univ. of Quebec at Montreal, Montreal, QC H3C 3P8, Canada, t16025@internet.uqam.ca)

The habituation techniques used in infant speech perception studies are based on the fact that infants show renewed interest towards novel stimuli. Recent work has shown the possibility of using artificial neural networks to model habituation and dishabituation (e.g., Schafer and Mareschal, 2001). In our study we examine whether the self-organizing-feature-maps (SOM) (Kohonen, 1989) are appropriate for modeling short-term habituation to a repeated speech stimulus. We found that although SOMs are particularly useful for simulating categorization, they can be modified to model habituation and dishabituation, so that they can be applied to direct comparisons with behavioral data on infants' speech discrimination abilities. In particular, we modified the SOMs to include additional parameters that control the relation of input similarity, lateral inhibition, and local and lateral activation between neurons. Preliminary results suggest that these parameters are sufficient for the network to simulate the loss of sensitivity of the auditory system due to the presentation of multiple tokens of a speech stimulus, as well as to model the recovery of sensitivity to a novel stimulus. The implications of this approach to infant speech perception research will be considered.

3:15

2pSC7. Phonetic representation of frequent function words in 8-month-old infants. Rushen Shi (Dept. of Psych., Univ. of Quebec at Montreal, Montreal, QC H3C 3P8, Canada, shi.rushen@uqam.ca), Janet Werker (Univ. of BC, Vancouver, BC V6T 1Z4, Canada), and Anne Cutler (Max Planck Inst. for Psycholinguist., Nijmegen, The Netherlands)

Recent work by a number of researchers showed that even preverbal infants detect and recognize functors in continuous speech. In our research, English-learning infants aged 11 to 13 months, but not 8 months, recognized frequent and infrequent functors as a class, and represented them in segmental detail (Shi *et al.*, 2003; Shi *et al.*, 2004). Here we report a study on 8-month-old infants' recognition and representation of high versus low frequency functors. Infants heard sequences containing a lexical word preceded by a high frequency functor "the," versus a nonsense functor "kuh," differing from "the" only in the initial consonant, with the prosody unchanged. Another group of 8-month-olds heard sequences containing a lexical word preceded by a low frequency functor "its," versus a nonsense functor "ots." Recognition of functors would be indicated by longer listening time to sequences containing real functors. Results reveal no differential listening time between "the+lexical word(LW)" and "kuh+LW," nor between "its+LW" and "ots+LW;" however, "the+LW" and "kuh+LW" together induced longer listening time than "its+LW" and "ots+LW." We conclude that 8-month olds recognize the frequent, familiar "the" in continuous speech, but it is underspecified phonetically in infants' initial lexicon. Our previous work indicates detailed specification by 11 months.

3:30

2pSC8. Segments and segmental properties in cross-language perception: Korean perception of English obstruents in various prosodic locations. Kenneth de Jong, Noah Silbert, and Hanyong Park (Dept. of Linguist., 322 Memorial Hall, Indiana Univ., Bloomington, IN 47405, kdejong@indiana.edu)

Experimental models of cross-language perception and second-language acquisition (such as PAM and SLM) typically treat language differences in terms of whether the two languages share phonological segmental categories. Linguistic models, by contrast, generally examine properties which cross classify segments, such as features, rules, or prosodic constraints. Such models predict that perceptual patterns found for one segment will generalize to other segments of the same class. This paper presents perceptual identifications of Korean listeners to a set of

voiced and voiceless English stops and fricatives in various prosodic locations to determine the extent to which such generality occurs. Results show some class-general effects; for example, voicing identification patterns generalize from stops, which occur in Korean, to nonsibilant fricatives, which are new to Korean listeners. However, when identification is poor, there are clear differences between segments within the same class. For example, in identifying stops and fricatives, both point of articulation and prosodic position bias perceptions; coronals are more often labeled fricatives, and syllable initial obstruents are more often labeled stops. These results suggest that class-general perceptual patterns are not a simple consequence of the structure of the perceptual system, but need to be acquired by factoring out within-class differences.

3:45

2pSC9. Cross-language perceptual category mapping: Korean perception of English obstruents. Hanyong Park, Kenneth de Jong, and Noah Silbert (Dept. of Linguist., 322 Memorial Hall, Indiana Univ., Bloomington, IN 47405, hanyongpark@indiana.edu)

Models, such as SLM and PAM, predict that performance on second language sounds is determined by the perceptual relationship of the sounds to the original language categories. To measure this relationship, Schmidt [J. Acoust. Soc. Am. **99**, 3201–3211] had native Korean speakers classify English consonant productions into Korean orthographic categories, and assess the similarity of the consonants to the chosen categories. The current experiment further examines how Korean labeling relates to accuracy in using English orthographic categories. Results show Koreans poorly identify sounds rated as dissimilar from Korean categories. Similarly, sounds that are inconsistently labeled with Korean labels are less accurately identified. These results suggest that accuracy varies with the nearness of the English and a Korean category, and thus English categories are developed from original Korean categories. However, other results indicate that sounds straddling Korean categories can be very accurately labeled, suggesting a complete suppression of native contrasts. In addition, identification accuracy for subjectively odd sounds can be very high, suggesting the development of a new category. Similarly, cases in which both Korean and English labeling is inconsistent show no relationship between the Korean and the English labels, indicating that the English categories are constructed apart from the existing Korean categories.

4:00

2pSC10. Production of Mandarin tones by 3-year-old children acquiring L1 (Mandarin) in an L2 (English) environment. Pui-san Wong and Richard G. Schwartz (Ph.D. Prog. in Speech and Hearing Sci., The Grad. School and Univ. Ctr., City Univ. of New York, 365 Fifth Ave., New York, NY 10016)

Thirteen monolingual Mandarin-speaking children residing in the U.S. were recruited to examine their production of the four Mandarin lexical tones in monosyllabic words. A picture-naming task was used to elicit the children's productions of lexical tones in isolated words and in sentence final position. Four mothers were asked to say the same set of words to their children in a picture reading activity. The children's and the mothers' productions were recorded and low-pass filtered at 500 and 400 Hz, respectively, to eliminate phonemic and semantic information. Ten Mandarin-speaking judges were recruited to identify the children's and adults' tone productions from the filtered stimuli. Contrary to the findings of L1 research conducted in countries where Mandarin is the language of the environment, the present results revealed that the lexical tones produced by 3-year-old children acquiring Mandarin as their first language in the U.S. were not yet adultlike. Children's tone productions were more difficult to categorize than the mothers' productions. The judges had significantly more difficulty identifying children's dipping tones than the children's level tones, rising tones, or falling tones, suggesting that the dipping tone posed the most difficulties for the children.

2pSC11. Language preference in monolingual and bilingual infants. Ayasha Valji and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., 1266 Pine Ave., Montreal, QC H3G 1A8, Canada linda.polka@mcgill.ca)

Previous research shows that infants being raised in single-language families have some basic language discrimination abilities at birth, that these skills improve over the first 6 months of life, and that infants are attending to the rhythmic properties of language to perform these skills. Research has also revealed that newborns and older babies from monolingual families prefer listening to their native language over an unfamiliar language. Data on language discrimination and preference in bilingual infants is very limited but is necessary to determine if the patterns and rate of bilingual language development parallel those of monolingual development, or if exposure to more than one language modifies developmental patterns. The present study addresses this issue by comparing language preference in monolingual English, monolingual French, and bilingual English–French infants between 3 and 10 months of age. Infant preference to listen to passages in three rhythmically different languages (English, French, Japanese) was assessed using a visual fixation procedure. Passages were produced by three female native speakers of each language. Findings will show how native language preference is affected by age and language experience in infants who experience monolingual and bilingual language exposure.

2pSC12. Syllable-final fricatives: Dutch and English listeners' processing of voicing. Mirjam Broersma (Max Planck Inst. for Psycholinguist., Postbus 310, 6500 AH Nijmegen, The Netherlands, mirjam.broersma@mpi.nl)

English and Dutch have both voiced and voiceless fricatives, and, in English, both occur in either syllable-initial or -final position. In Dutch, however, only voiceless fricatives can occur in syllable-final position. A categorization experiment investigated the processing of the voicing distinction in English fricatives by Dutch and English listeners, and in particular whether Dutch listeners can distinguish voicing contrasts in syllable-final position, and whether preceding vowel length informs their voicing judgments. Dutch and English participants heard tokens from an 11-step /v-/f/ continuum, at the end of a single nonword. As the nonword context was kept constant within each block, vowel length could not serve as a cue. Half of the participants heard nonword contexts which were originally pronounced with a final /v/, the other half with final /f/. Therefore, in both cases a mismatch occurred for a subset of the items between vowel length and the other information in the signal. This mismatch hampered the performance of the English but not the Dutch listeners, so that the Dutch listeners outperformed the English in mismatching cases. This suggests that the Dutch did not rely on vowel length as a cue to voicing as strongly as the English.

4:45–5:15

Panel Discussion

TUESDAY AFTERNOON, 25 MAY 2004

RIVERSIDE SUITE, 1:25 TO 4:15 P.M.

Session 2pSP

Signal Processing in Acoustics and Underwater Acoustics: Acoustic Communications—Algorithms and Performance Analysis

Milica Stojanovic, Chair

Massachusetts Institute of Technology, Cambridge, Massachusetts 02139

Chair's Introduction—1:25

Contributed Papers

1:30

2pSP1. Performance predictions for acoustic communication using passive phase conjugation. Daniel Rouseff (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Passive phase conjugation processing for acoustic communication combines temporal matched filtering with spatial diversity. An efficient decision-directed procedure has been implemented to calculate the matched filters and has been applied successfully to experimental data. In the present paper, a model is developed for the expected mean-squared error in the demodulation output. A normal mode expansion is used for the phase-encoded acoustic field. The matched filter integration is evaluated using a stationary phase approximation. This intermediate result is interpreted using the waveguide invariant. After combining across the array and sampling, formal statistical averages of the demodulation output are calculated. Results are parametrized in terms of the number of array elements and their spatial distribution. The effects of truncating the matched filters and reducing the bandwidth are also quantified. Performance predictions are shown to be in good agreement with experimental results. [Work supported by ONR under the ARL program.]

1:45

2pSP2. Performance comparisons between passive-phase conjugation and decision-feedback equalizer for underwater acoustic communications. T. C. Yang (Naval Res. Lab. Code 7120, 4555 Overlook Ave., Washington, DC 20375)

Passive-phase conjugation (PPC) uses passive time reversal to remove intersymbol interferences (ISI) for acoustic communications in a multipath environment. It is based on the theory of signal propagation in a waveguide, which says that the Green function (or the impulse response function) convolved with its time-reversed conjugate, summed over a large aperture vertical array of receivers is a delta function in space and time. A decision feedback equalizer (DFE) uses a nonlinear filter to remove ISI based on the minimum mean square errors (MSE) between the estimated and the true (or decision) symbols. These two approaches are motivated by different principles. In this paper, we analyze both using a common framework, illustrating the commonality and differences, pro and con between the two methods, and compare their performances in realistic ocean environments. The performance measures are MSE, output signal-to-noise ratio and bit error rate (BER) as a function of the number of receivers. For a small number of receivers, DFE outperforms PPC in all measures. As the

number of receivers increases the BER for both processors approaches zero, but at a different rate. The results are supported by both simulated and real data. [Work supported by ONR.]

2:00

2pSP3. Robust adaptive decision feedback equalization using residual prediction errors. James C. Preisig (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Channel estimated based adaptive decision feedback equalization (DFE) relies on estimates of the channel impulse response and interfering noise field to calculate optimal equalizer filter coefficients. While the DFE offers superior performance to either passive time-reversal or minimum mean squared error (mmse) linear equalization when good quality channel estimates are available, the DFE is more sensitive than the other two techniques to channel estimation errors. A technique incorporating the residual prediction error of the channel estimation algorithm is presented. This algorithm, referred to as the residual prediction error DFE (RPE-DFE), maintains the superior performance that is characteristic of the DFE operating with good quality channel estimates while improving the robustness of the DFE to channel estimation errors. Experimental data is presented that allows for a direct comparison of the RPE-DFE performance with the performance of the traditional DFE, the passive time reversal and mmse linear equalizers, and robust DFEs proposed earlier [M. Stojanovic, J. Proakis, and J. Catipovic, IEEE T-Comm. **43**, 877–886 (1995)]. [Work supported by ONR Ocean Acoustics.]

2:15

2pSP4. Information and data real time transmission acoustic underwater system: TRIDENT. Joel Trubuil, Joel Labat (ENST Bretagne Technopole Brest Iroise CS 83818, 29238 Brest, Cedex 3, France, joel.trubuil@enst-bretagne.fr), and Gerard Lapierre (GESMA, BP42 29240, Brest Naval, France)

The objective of the Groupe d'Etudes Sous-Marines de l'Atlantique (GESMA) is to develop a robust high data rate acoustic link. A real-time receiver recently developed at ENST Bretagne has just been designed to cope with all perturbations induced by such harsh channels. In order to cope with channel features, a spatio-temporal equalizer introduced by J. Labat *et al.* [Brevet FT no. 9914844, "Perfectionnements aux dispositifs d'egalisation adaptative pour recepteurs de systemes de communications numriques," Nov. 1999] was recently implemented and evaluated. This equalizer is the core of the receiver platform [Trubuil *et al.*, "Real-time high data rate acoustic link based on spatio temporal blind equalization: the TRIDENT acoustic system," OCEANS 2002]. This paper provides an overview of this project. The context of the study and the design of high data rate acoustic link are presented. Last Brest harbor experiments (2002, 2003) are described. The real time horizontal acoustic link performances are evaluated. Two carriers frequencies are available (20, 35 kHz). Acoustic communications for bit rate ranging from 10 to 20 kbps and for channel length (shallow water) ranging from 500 to 4000 m have been conducted successfully over several hours.

2:30

2pSP5. Reliable Doppler-spread estimation using an evolutionary algorithm for underwater acoustic communications over channels with frequency-selective fading. Jasdeep S. Dhanoa, Richard F. Ormondroyd, and Evan J. Hughes (Dept. of Aerosp., Power and Sensors, Cranfield Univ., The Royal Military College of Sci., Shrivenham, Oxfordshire SN6 8LA, UK)

There is a pressing need for reliable wideband underwater acoustic communication for enhanced performance of diver communications, underwater networks, control links to ROVs and data links to remote sensors and AUVs. The performance of such links is compromised by the doubly-spread acoustic channel, where multipath time-delay spread causes intersymbol interference due to frequency-selective fading and Doppler-spread results in additional signal distortion and temporal variation of the channel characteristics. However, if the channel characteristics can be estimated

accurately, especially the Doppler-spread, many of these problems can be resolved. Doppler-spread occurs because the Doppler velocity may be different on each path. The paper describes a method for improved-reliability Doppler spread estimation of channels that experience frequency-selective fading. The method uses a wideband sounding signal to overcome the frequency selective fading and the estimation of the Doppler spread is obtained from this complex waveform using multi-layered optimization based on evolutionary algorithms. The method has a wide dynamic range and can estimate the Doppler shift of weak multipath components in the presence of strong multipath components. The accuracy of the Doppler-spread estimate is not limited by the short duration of sounding signal and the method performs well even in the presence of significant noise.

2:45–3:00 Break

3:00

2pSP6. Design of a symmetric multicarrier modulation for underwater acoustic communications. Yi Wang (Information and Coding Theory Lab., Univ. of Kiel, Kaiserstr. 2, D-24143 Kiel, Germany, yw@tf.uni-kiel.de)

Mobile underwater acoustic (UWA) channels always exhibit large delay spread and Doppler spread. Multicarrier modulation is an efficient technique for the communications over multipath channels. However, the orthogonality of the transmit signals, e.g., OFDM signals, will be destroyed by the Doppler spread and interference will be produced. Thus, an equalizer is required in the receiver. For this reason, we argue that multicarrier modulation with Gaussian pulse (symmetric in time and frequency domains) as the transmit filter is more suitable for the UWA communication systems than the OFDM. The key problem for the multicarrier modulation is to design/optimize the number of carriers and the parameter alpha in the Gaussian pulse which will greatly affect the system performance. In this paper, we investigate the optimization problem for four kinds of channels: AWGN channel, frequency-selective channel, time-selective channel and doubly selective channel. Channel information is assumed known in the transmitter by feedback. Analytical results are obtained and the numerical examples are given. To achieve the cost function, a 2D MMSE-DFE equalizer is used. Finally, we propose the balanced design for the practical engineering.

3:15

2pSP7. High-rate acoustic link for underwater video transmission using differential amplitude phase shift keying (DAPSK). Costas Pelekanakis, Milica Stojanovic (MIT, 292 Main St., Cambridge, MA 02139, gas@mit.edu), and Lee Freitag (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

A high bit-rate acoustic link for underwater video transmission is examined. Currently, video encoding standards allow video transmission at bit rates on the order of 64 kbps. To provide an acoustic transmission capability that meets this bit-rate requirement, we focus on the use of high-level bandwidth-efficient modulation methods. We consider the use of 16-, 32-, and 64-differential amplitude and phase shift keying (DAPSK) with varying number of amplitude levels. DAPSK does not require explicit carrier phase synchronization at receiver, but instead relies on differentially coherent detection. It thus represents an alternative to conventional quadrature amplitude modulation (QAM) methods, whose high-level constellations are sensitive to phase distortion induced by the channel. System performance is demonstrated experimentally, using 25 000 symbols/s at a carrier frequency of 75 kHz over a short vertical path. Excellent results were obtained, thus achieving bit rates as high as 150 kbps, which are sufficient for real-time transmission of compressed video.

3:30

2pSP8. Underwater acoustic communication experimentation: The acoustic communications and data storage (ACDS) system. M. McCord, J. Schindall, T. C. Yang, P. Gendron, and W.-B. Yang (Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375)

The Naval Research laboratory has developed a sea-going system, currently consisting of three acoustic communications and data storage (ACDS) buoys. The buoys can be deployed in the water column, moored to the ocean floor, or towed behind a ship to emulate an autonomous underwater vehicle (AUV). The system is intended for semi-real-time acoustic communications *in situ* and continuous recording of the raw acoustic data in the water for postexperiment analysis. The purpose is to study environmental issues affecting high data rate point-to-point acoustic communications as well as multiple-in-multiple-out acoustic networking. Each ACDS has an acoustic projector and an array of eight hydrophones. It is designed for two frequency bands: 2–5 kHz and 15–25 kHz. The system has been deployed in several oceans. Modulation signals used include binary and quadrature phase-shifted keying (BPSK/QPSK), frequency-hopped frequency-shifted keying (FH FSK), orthogonal frequency division multiplexing (OFDM) and code division multiple access (CDMA) signals. The acoustic modem is based on the Acoustic Modem Software developed by WHOI and improved by the Naval Underwater Warfare Center, Newport, RI. Acoustic networking signals are used to aid vehicle navigation. [Work supported by ONR.]

3:45

2pSP9. Doppler compensated underwater acoustic communication system. Anand Raj, Binu George, M. H. Supiya, James Kurian, and P. R. Saseendran Pillai (Dept. of Electron., Cochin Univ. of Sci. and Technol., Cochin 682 022, India)

Spread spectrum methods are used in communication systems to provide a low probability of intercept in hostile environments and multiple access capability in systems shared by many users as well as to provide high processing gain in channels where the transmitted signal is distorted

by multipath effects. Such systems serve to be an effective tool for underwater telemetry environments, where multipath propagation effect and Doppler spreading is seen to be more predominant. This paper describes the implementation of a Doppler compensated underwater telemetry system based on CDMA technique. The system consists of multiple CDMA transmitters and a phase locked loop based carrier recoverable CDMA receiver. The effects of the Doppler shift can be compensated by the carrier recovery subsystem in the demodulator, based on PLL technique, which extracts the carrier frequency/phase and simultaneously demodulates the signal. The decision device in the receiver consists of a PN sequence generator as well as a bank of correlators, which are used to determine the data transmitted. The system simulation has been implemented in MATLAB. The advantage of this system is that multiple transmitting stations can transmit simultaneously to a central receiver, thereby increasing the system throughput.

4:00

2pSP10. Environmental and motion effects on orthogonal frequency division multiplexed signaling. Paul J. Gendron and T. C. Yang (4555 Overlook Ave. SW, Washington, DC 20375)

The use of tightly spaced orthogonal tones for communication through shallow water allows for diverse modulation strategies, from moderately efficient noncoherent on-off keying to fully coherent signaling. Critical to the effectiveness of orthogonal frequency division multiplexed (OFDM) schemes is the estimation of both source dilation, due to source-receiver relative motion, and the channel transfer function. A dilation process estimator based on a spline model of dilation is presented and when coupled with decision directed channel estimation communication reliability is improved. The method is tested on OFDM signaling schemes at 18 kHz center frequency and 4 kHz bandwidth under diverse source motion scenarios and ranges during MREA(03) experiments off the north coast of Elba, Italy. Channel probing experiments characterize the environment in terms of coherence time, multi-path delay and angle spread. The quantitative effect of frequency selectivity and coherence time on the performance of OFDM signaling is addressed. Bit error rates are reported. [Work supported by ONR.]

2p TUE. PM

TUESDAY AFTERNOON, 25 MAY 2004

VERSAILLES BALLROOM, 1:25 TO 3:30 P.M.

Session 2pUW

Underwater Acoustics: Ambient Noise

Anthony J. Eller, Chair

Science Applications International Corp., 1710 SAIC Drive, McLean, Virginia 22102

Chair's Introduction—1:25

Contributed Papers

1:30

2pUW1. Using ocean ambient noise for array element localization. Karim G. Sabra, Philippe Roux, W. A. Kuperman, W. S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093, ksabra@mpl.ucsd.edu), and Gerald L. D'Spain (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0704)

Time delays, associated with the direct and surface reflected arrivals, between the elements of a bottom hydrophone array can be extracted using ambient noise cross correlations [Sabra *et al.*, *J. Acoust. Soc. Am.* **114**, 2462 (2003)]. This is confirmed using long-time noise recordings that were collected in May 1995 near the S. California coast at an average depth of 21 m. The noise is mainly biological in the frequency range of

350–700 Hz [D'Spain *et al.*, *J. Acoust. Soc. Am.* **99**, 2453 (1996)]. These coherent wavefront arrivals across the array's aperture are used for array element localization.

1:45

2pUW2. Testing of ambient noise forecasting algorithms with LADC noise data. Anthony I. Eller (Sci. Applications Intl. Corp., 1710 Saic Dr., McLean, VA 22102, anthony.i.eller@saic.com), George E. Ioup, Juliette W. Ioup, and James P. Larue (Univ. of New Orleans, New Orleans, LA 70148)

Long-term, omnidirectional underwater acoustic noise was measured by three buoys in the northern Gulf of Mexico over a 36-day period during the summer of 2001 as part of the Littoral Acoustic Demonstration Center

project, phase I. This extensive data set is used to develop and evaluate algorithms for near-term temporal prediction of noise, based on immediate *in situ* data. Noise time series data in one-third-octave bands from 10 to 5000 Hz are processed to extract characterizing parameters such as mean level, variance, and temporal coherence across all frequency bands. The fluctuation spectrum is analyzed to determine how the noise variability is distributed with temporal period, and eigenvectors based on the fluctuation covariance matrix are used to identify underlying basic properties of noise variation. These tools are then used to predict future values of noise based on measurements at preceding times. [Research supported by ONR.]

2:00

2pUW3. Vertical directivity of ambient noise in the presence of internal waves. Richard B. Evans (Sci. Applications Intl. Corp., 23 Clara Dr., Mystic, CT 06355, richard.b.evans@saic.com)

Near surface sources of ambient noise, such as shipping and wind, excite only steeply traveling sound waves. After a sufficient range, the steepest are stripped away by loss mechanisms, such as bottom interaction. Consequently, the vertical directivity of surface generated noise is expected to be peaked about two (up- and down-going) angles with very little contribution near the horizontal. This expectation is not always met in measurements. A possible explanation, for the noise at the horizontal, is the random scattering of the up- and down-going waves by volume inhomogeneities caused by internal waves. The purpose of this paper is to demonstrate the validity of this hypothesis. The coupled power theory of Dozier and Tappert [J. Acoust. Soc. Am. **63**, 353–365 (1978)], in an updated form, is applied to the calculation of noise due to distant near surface sources. The vertical directivity of the resultant field is shown to demonstrate that internal wave scattering can produce noise at the horizontal. Rough surface scattering is expected to have a similar effect.

2:15

2pUW4. Ambient noise effects on towed-array search optimization. Donald R. DelBalzo and Erik R. Rike (Neptune Sci., Inc., 40201 Hwy. 190 E., Slidell, LA 70461)

Oceanographic and acoustic conditions in littoral environments are extremely complex and dynamic. Spatial and temporal variability of low-frequency signal and noise fields destroys the basic homogeneous assumptions associated with standard tactical search concepts, like simple ladder patterns. Genetic algorithms have recently been applied to the signal part of this problem [D. P. Kierstead and D. R. DelBalzo, "A genetic algorithm approach for planning search paths in complicated environments," Military Operations Research Journal (March/April 2003)] to produce near-optimal, nonstandard search tracks that maximize probability of detection in uniform noise fields. In a complex environment, an optimal search plan will also depend on the spatial and temporal properties of AN. The dynamic ambient noise model (DANM) was used to produce low-frequency directional noise fields, based on discrete ship tracks. Then, optimal genetic search tracks were calculated for towed arrays, in the midst of spatial variability and temporal fluctuations of noise. The results show the importance of careful consideration of AN when designing optimum tactics. [Work sponsored by NAVSEA under the LCS project.]

2:30

2pUW5. An iso-deviant strategy for efficient ambient noise predictions with EAGLE. Erik R. Rike and Donald R. DelBalzo (Neptune Sci., Inc., 40201 Hwy. 190 E., Slidell, LA 70461, delbalzo@neptunesci.com)

Transmission loss (TL) computations in littoral areas require a dense spatial and azimuthal grid to achieve acceptable accuracy and detail, which is very slow. This problem of accuracy versus speed led to a new concept, OGRES (Objective Grid/Radials using Environmentally sensitive Selection), which produces sparse, irregular acoustic grids, with controlled accuracy. Recent work to further increase accuracy and efficiency with better metrics and interpolation led to EAGLE (Efficient Acoustic Gridder for Littoral Environments). On each iteration, EAGLE produces an acous-

tic field with approximately constant spatial uncertainty (hence, iso-deviance), yielding TL predictions with ever-increasing resolution and accuracy. This work adapts EAGLE to ambient noise computations. The Dynamic Ambient Noise Model (DANM) allows accurate, detailed estimation of the mean and variance of ambient noise in both space and time, but its TL computations are too slow for many applications. In the present work, a series of EAGLE acoustic field predictions was used by DANM (and compared to the dense full-grid solution) to determine the relationship between transmission loss uncertainty and noise-field uncertainty for a complex littoral area. An example is presented where approximately an order of magnitude efficiency improvement (over regular grids) is demonstrated. [Work sponsored by ONR under the LADC project.]

2:45

2pUW6. Ambient noise analysis of deep ocean measurements. Roy D. Gaul (Blue Sea Corp., Ste. 515, 14300 Cornerstone Village Dr., Houston, TX 77014), David P. Knobles (Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029), and A. F. Wittenborn (TRACOR Sci. and Systems)

In October 1975 a measurement exercise designated CHURCH OPAL was done in the northeast Pacific Ocean to assess undersea acoustic noise and propagation phenomena. In 2003 the 10 days of deep ocean multiple hydrophone recordings during CHURCH OPAL were recovered and digitized. This paper presents results previously reported but unavailable for general distribution. The earlier work is augmented with more complete and detailed analyses using modern analytical techniques. Particular attention is given to statistical characterization of ambient noise within and beneath the deep sound channel in relation to distant shipping and local wind speed. [Work supported by ONR.]

3:00

2pUW7. Analysis on vertical directivity of shallow-water ambient noise in South China Sea. Ruey-Chang Wei (Inst. of Undersea Technol., Natl. Sun Yat-sen Univ., 70, Lien-hai Rd., Kaohsiung, Taiwan, rcwei@mail.nsysu.edu.tw), Hsiang-Chih Chan (Natl. Taiwan Univ., Taipei 804, Taiwan), and Po-Chang Lin (Natl. Sun Yat-sen Univ., Kaohsiung, Taiwan)

Beamforming is used to analyze the vertical directivity of the low-frequency ambient noise in shallow water. The data were measured by the vertical line array (VLA) of the Asian Seas International Acoustics Experiment (ASIAEX), from 3 May to 16 May 2001, and three frequencies, 200, 400, and 800 Hz, were studied. The results of the calculation show that the sound source of 200 and 400 Hz is from the horizontal direction and contributed by distant shipping, and the source of 800 Hz is mainly from high grazing angle that generated from surface noise. The well-known noise notch was also found in the study; the low-angle energy reflections from surface and bottom are the function of frequency, sound-speed profile, bathymetry, and the noise distribution. The time fluctuation of the noise notch is shown and correlated with oceanographic changes. The influences of internal wave and pass typhoon Cimaron during the experiment are significant too on vertical directivity of ambient noise.

3:15

2pUW8. Initial study of ambient noise modeling for 2001 ASIAEX SCS Experiment. Ruey-Chang Wei (Inst. of Undersea Technol., Natl. Sun Yat-sen Univ., 70, Lien-hai Rd., Kaohsiung, Taiwan, rcwei@mail.nsysu.edu.tw), Chi-Fang Chen (Natl. Taiwan Univ., Taipei, 804, Taiwan), and Hsiang-Chih Chan (Natl. Taiwan Univ., Taipei 804, Taiwan)

Although ambient noise has been an object of study for a long time, there is little modeling done for real ocean environment to reflect oceanographic changes. The numerical simulation in this study is based on PE code, with environmental and bathymetry data from the 2001 ASIAEX SCS experiment. Simple sources were distributed near sea surface in the model, and source levels in significant range were estimated by the wave height data of experiment. It is well known that strong internal waves

often occur in the South China Sea, which can cause significant change in water temperature distribution, thus the sound propagation. Moreover, the roughness of sea surface and bottom are also important factors in noise energy distribution. In this initial study, the fluctuation of noise energy

would be evaluated by the varying spatial ocean environment. The computational noise levels would be compared with measurements on vertical line array in ASIAEX SCS experiment for adjusting the parameters and source levels.

TUESDAY AFTERNOON, 25 MAY 2004

CONFERENCE ROOM L, 2:00 TO 3:15 P.M.

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

to be held jointly with the

**ANSI-Accredited U.S. Technical Advisory Group (TAG) Meetings for:
ISO/TC 108, Mechanical Vibration and Shock
ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied
to machines, vehicles and structures
ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring instruments
ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
and
ISO/TC 108/SC 6 Vibration and shock generating systems**

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. There will be a report on the interface of S2 activities with those of ISO/TC 108 and its subcommittees, including plans for future meetings of ISO/TC 108 and/or its Subcommittees. The Technical Advisory Groups for ISO/TC 108 and the Subcommittees listed above consists of members of S2 and other persons not necessarily members of those Committees. Open discussion of committee reports is encouraged.

Scope of S2: Standards, specifications, methods of measurement and test, and terminology in the fields of mechanical vibration and shock, and condition monitoring and diagnostics of machines, but excluding those aspects which pertain to biological safety, tolerance and comfort.

TUESDAY AFTERNOON, 25 MAY 2004

CONFERENCE ROOM L, 3:30 TO 5:00 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

to be held jointly with the

**ANSI-Accredited U.S. Technical Advisory Group (TAG) Meeting for:
ISO/TC 108/SC 4, Human Exposure to Mechanical Vibration and Shock**

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. There will be a report on the interface of S3 activities with those of ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock, including plans for future meetings of ISO/TC 108/SC 4. The US Technical Advisory Group for TC 108/SC 4 consists of members of S3 and other persons not necessarily members of this Committee. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 8:00 a.m. on Tuesday, 25 May 2004.

Scope of S3: Standards, specifications, methods of measurements and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics, shock, and vibration which pertain to biological safety, tolerance, and comfort.

2p TUE. PM

Banquet

Dinner

6:30 p.m.–8:00 p.m.

Lawrence A. Crum and Anthony A. Atchley, Masters of Ceremonies

Program

Welcome Remarks by Ilene J. Busch-Vishniac, President

Presentation of Certificates to New Fellows

Abeer Alwan	Jeffrey A. Nystuen
David H. Chambers	Andrew J. Oxenham
Robin O. Cleveland	Christopher J. Plack
Li Deng	Daniel Rouseff
Alain de Cheveigne	Carrick Talmadge
Gary J. Heald	D. Keith Wilson
Jian Kang	Eric J. Woods

Viewing of Special Film made for the
75th Anniversary Celebration

Closing Remarks