The Journal of the Acoustical Society of America

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171st Meeting
Acoustical Society of America

Salt Lake Marriott Downtown at City Creek Hotel
Salt Lake City, Utah
23–27 May 2016

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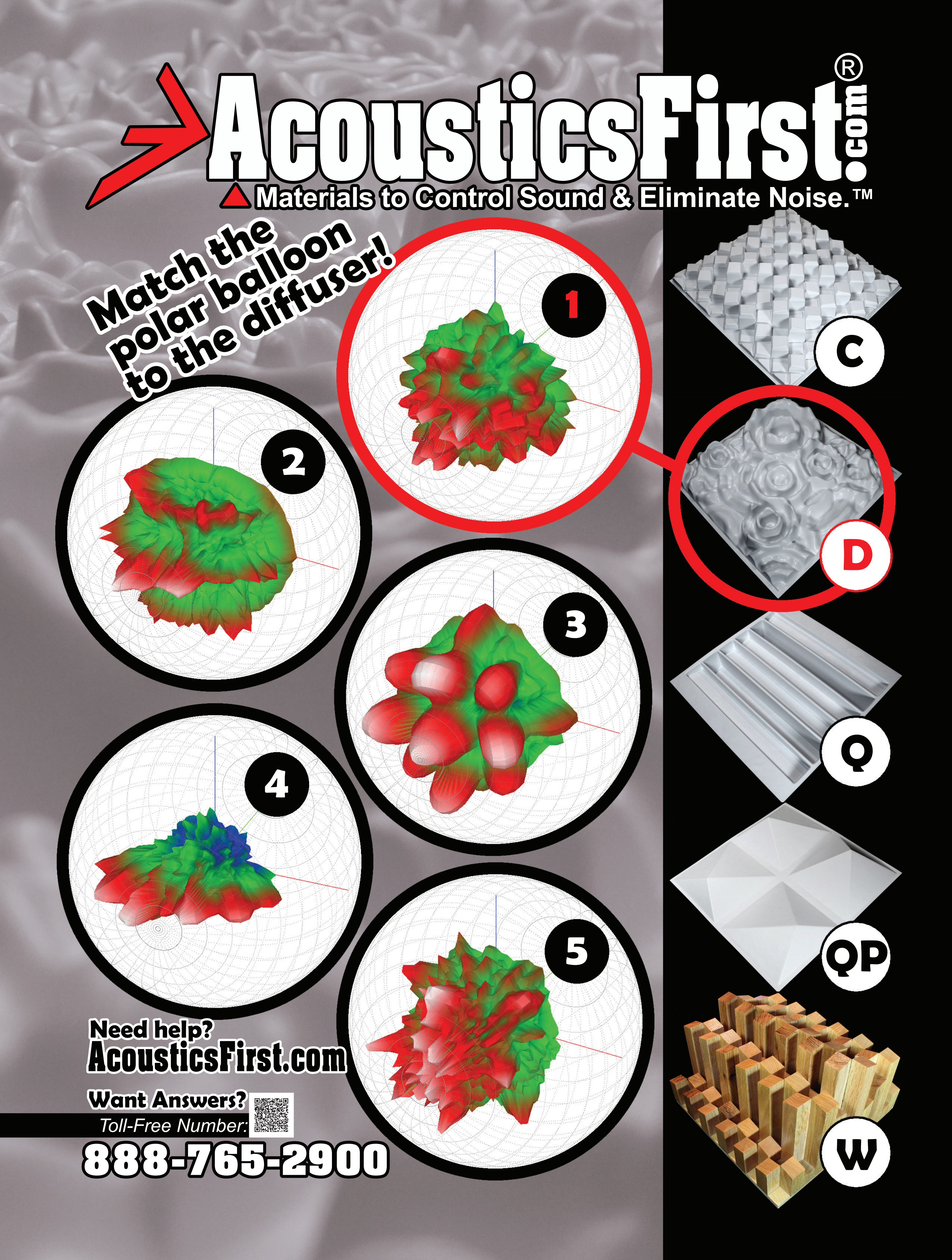
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The Journal of the Acoustical Society of America

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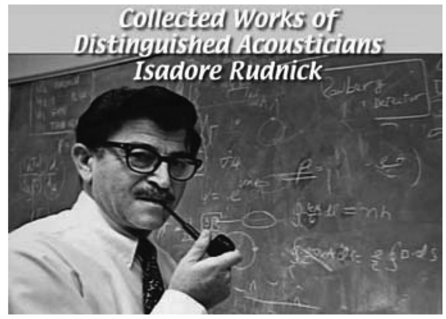
Collected Works of Distinguished Acousticians

Isadore Rudnick



The first in this series of the Collected Works of Distinguished Acousticians is that of Isadore Rudnick (May 8, 1917 - August 22, 1997). Rudnick was honored by the Acoustical Society of America (ASA) with the R. Bruce Lindsay (Biennial) Award in 1948, the Silver Medal in Physical Acoustics in 1975, and the Gold Medal in 1982. He was recognized for his acoustics research in low temperature physics with this field's most prestigious award, the Fritz London Memorial Award, in 1981 and was inducted into the National Academy of Science in 1983. Izzy's research in physical acoustics addressed boundary propagation, reciprocity calibration, high intensity sound and its biological effects, nonlinear sound propagation, and acoustics in superconductors and superfluids, including critical phenomena in bulk and thin films. The first disc in this three disc set contains reprints of Rudnick's papers from scientific journals, including 26 from the Journal of the Acoustical Society of America, and 87 from other prestigious journals, as well as some consulting reports and invited papers presented at international meetings which would otherwise be difficult to obtain. The second disc includes a montage of photographs of Rudnick with colleagues and family, Rudnick's prize winning film "The Unusual Properties of Liquid Helium", and a video of the Plenary session at the ASA's 100th meeting where Rudnick presented 90 minutes of unique and stage-sized acoustics demonstrations. While videotaped under poor conditions and of lamentable quality, the reprocessed video of acoustics demonstrations is one of the most valuable parts of this collection. The third disc is a video recording of the Memorial Session held at the 135th meeting of the ASA, which provides a comprehensive summary of Rudnick's contributions as described by former students and collaborators.

The CD was compiled by Julian D. Maynard and Steven L. Garrett of the Pennsylvania State University, State College, Pennsylvania.

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TECHNICAL PROGRAM SUMMARY
***Indicates Special Session**

Monday morning

*1aAA Classroom Acoustics
*1aEDa Hands-On Acoustics Demonstrations for Middle- and High-School Students
*1aEDb Strategies for Effectively Communicating Science to Policy Makers, the Media, and the Public
*1aNS Community Noise I
*1aPA Computational Methods in Physical Acoustics I
1aPP Psychological and Physiological Acoustics Potpourri (Poster Session)

Monday afternoon

*1pAA Sound System Design, Optimization, and Intelligibility
*1pAB Comparative Hearing - Honoring Dick Fay
*1pED Listen Up and Get Involved
*1pID Introductions to Technical Committees
*1pNS Community Noise II
*1pPA Computational Methods in Physical Acoustics II
*1pSAa Analysis of Vibration Based Musical Instruments
1pSAb General Topics in Structural Acoustics and Vibration
1pSC Acoustics and Perception of Speech (Poster Session)
*1pSP Small Unmanned Aerial Vehicle (UAV) Detection, Tracking, and Classification

Monday evening

*1eID Tutorial Lecture on Acoustic Metamaterials: From Theory to Practice

Tuesday morning

*2aAAa Forensic Studies from Noise Identification to Solutions in the Built Environment
*2aAAb Student Design Competition
*2aAO Acoustic Consistency of Ocean Models
2aBAa Acoustic Radiation Force and Elastography
2aBAb Therapeutic Ultrasound and Microbubbles
2aEA General Topics in Engineering Acoustics I
2aED Education in Acoustics General Topics
*2aMU Voice Registration in Amplified and Unamplified Singing
*2aNSa Noise Measurements With Mobile Apps
*2aNSb Noise and Vibration Impacts from Crossfit Training Facilities
*2aPA Vortex Beams and Radiation Torque Physics I
*2aPP Approaches to Improve Speech Understanding in Noise
*2aSC Intelligibility, Hearing Impairment, Aging (Poster Session)
*2aSP Acoustic Array Systems and Signal Processing I. Solitude
2aUW Target Physics and Scattering

Tuesday afternoon

*2pAA Predicting and Controlling Heating, Ventilating, and Air Conditioning Systems Noise
2pABa Cetacean Bioacoustics
2pABb Animal Bioacoustics Poster Session
*2pBA Biomedical Acoustics Student Paper Competition (Poster Session)
2pEA General Topics in Engineering Acoustics II
*2pMU Pitch, Dynamics, and Vowel Tuning in Choral Voice
*2pNS Statistical Learning Techniques in Noise Research
*2pPA Vortex Beams and Radiation Torque Physics II
*2pPPa Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical and Physiological Collaborations
*2pPPb Auditory Neuroscience Prize Lecture
*2pSAa Real-World Instructive Case Studies in Structural Acoustics and Vibration
*2pSAb Wavenumber Transform Methods
*2pSC Intelligibility Challenges: Speakers, Listeners, and Situations
*2pSP Comparison of Beamforming, Matched-Field Processing, and Time Reversal Techniques. Solitude

Wednesday morning

*3aAA Relating Perception to Room Acoustics Measurements and Metrics in Performing Arts Venues I
*3aAB Effects of Noise on Animals
*3aBA Controlled Drug Delivery and Release with Focused Ultrasound
3aED Education in Acoustics: Student Posters
*3aMU Teaching Musical Acoustics Courses and Laboratories at Any Level. Solitude
*3aPA Atmospheric Acoustic Phenomena I
*3aPP Quantitative Methodology in Both Physiological and Psychophysical Data Analysis Workshop
*3aSA Computational Methods in Structural Acoustics and Vibration
*3aSC Gender Effects in Speech Production and Perception
*3aSP Acoustic Array Systems and Signal Processing II
*3aUW Sediment Characterization Using Direct and Inverse Techniques I

Wednesday afternoon

*3pAA Relating Perception to Room Acoustics Measurements and Metrics in Performing Arts Venues II
3pAB Airborne/Automatic Animal Bioacoustics
3pBA Focused Ultrasound for Brain Treatments
*3pID Hot Topics in Acoustics
*3pPA Atmospheric Acoustic Phenomena II
*3pPP Beyond the Audiogram: Influence of Supra-Threshold Deficits
*3pSC Variation and Gender Effects (Poster Session)
*3pUW Sediment Characterization Using Direct and Inverse Techniques II

Thursday morning

*4aAA Emerging Parametric/Generative Design Tools in Architectural Acoustics
*4aAO Noise Impacts from the Industrialization of the Outer Continental Shelf and High Seas
*4aNS Wind Turbine Noise I
*4aPA Multiple Scattering I
4aPP Temporal Aspects of Auditory Processing (Poster Session)
*4aSAa Building Isolation from Seismic and Groundborne Vibration
*4aSAb Nuclear-Powered Thermoacoustics
4aSC Non-Native Speech Perception and Production (Poster Session)
*4aSP Detection and Estimation in Uncertain Acoustic Environments I
*4aUW Sediment Characterization Using Direct and Inverse Techniques III

Thursday afternoon

*4pAA Opera Rehearsal and Performance Spaces
*4pAO Acoustical Oceanographic Tools for the Study of Marine Ecosystems
4pBA Imaging
*4pMU Session in Honor of William J. Strong
*4pNS Wind Turbine Noise II
4pPAa General Topics In Physical Acoustics I
*4pPAb Multiple Scattering II
4pPP Lessons from Interrupted Speech: Methods and Models
*4pSC Constraints, Strategies, and Economies of Effort in Speech Production: Joseph S. Perkell's Contribution to Speech Science
*4pSP Detection and Estimation in Uncertain Acoustic Environments II
4pUW Acoustic Propagation in the Ocean

Friday morning

5aAA A Variety of Interesting Research and Observations in Architectural Acoustics
5aMU General Topics in Musical Acoustics. Solitude
5aNS Topics in Noise Control
5aPA General Topics In Physical Acoustics II
5aPP Spatial Hearing
5aSCa Phonetics of Under-Documented Languages I
5aSCb Phonetics of Under-Documented Languages II (Poster Session)
5aSCc Speech Production (Poster Session)
5aSP General Topics in Signal Processing
5aUW Underwater Noise

SCHEDULE OF STARTING TIMES FOR TECHNICAL SESSIONS AND TECHNICAL COMMITTEE (TC) MEETINGS

	M am	M pm	M eve	Tu am	Tu pm	Tu eve	W am	W pm	W eve	Th am	Th pm	Th eve	Fri am
SALON A	1aEDb 10:00	1pID 2:15		2aEA 8:00	2pEA 1:10 TCEA 4:30		3aUW 8:00	3pUW 1:30		4aUW 8:00	4pUW 1:30		5aNS 8:30
SALON B/C	1aNS 8:45	1pNS 1:30	1eID 7:00	2aMU 8:00	2pMU 1:00 2pSAb 3:40	TCPP 7:30	3aAA 8:30	3pAA 1:15	TCMU 7:30	4aNS 8:00	4pNS 1:30		5aSCa 8:00
SALON D	1aEDa 9:00	1pED 5:30		2aPP 8:00	2pPPa 1:00		3aPP 7:55	3pID 1:30			4pSC 1:30	TCNS 7:30	5aPP 7:55
SALON E		1pAA 1:00		2aAAa 8:00	2pSAa 1:00		3aSP 8:30			4aSC 8:00	4pAA 1:00		5aSCc 9:00
SALON F		1pSC 1:00		2aSC 8:00 2aAAb 8:30	2pBA 1:00		3aSC 8:00			4aPP 8:00	4pPAa 1:00		5aSCb 10:05
SALON E/F	1aPP 8:00												
SALON G		1pAB 1:30		2aNSa 8:00 2aNSb 10:40	2pNS 1:30	TCSA 7:30	3aED 8:00	3pSC 1:00	TCSP 7:30	4aAA 8:30	4pBA 1:30	TCSC 7:30	5aSP 8:00
SALON H	1aPA 8:15	1pPA 1:30		2aPA 8:00	2pPA 1:00 2pPPb 4:25	TCPA 7:30	3aBA 8:15	3pBA 1:00	TCBA 7:30	4aPA 8:00	4pPAb 1:00		5aPA 8:00
SALON I	1aAA 9:00	1pSP 1:30		2aAO 8:00 2aED 10:25	2pABa 1:00 2pABb 3:30	TCAB 7:30	3aAB 8:15	3pAB 1:30		4aAO 8:00	4pAO 1:00		5aAA 8:00
SALON J		1pSAa 1:00 1pSAb 2:40		2aUW 8:30	2pSC 2:10	TCAO 7:30	3aPA 8:30	3pPA 1:15		4aSAa 8:30 4aSAb 10:50	4pPP 1:30	TCUW 7:30	5aUW 8:30
SNOWBIRD/ BRIGHTON				2aBAa 8:00 2aBAb 10:00	2pAA 1:00	TCAA 7:30	3aSA 8:30	3pPP 1:00		4aSP 10:30	4pSP 1:30		
SOLITUDE				2aSP 8:00	2pSP 1:00		3aMU 9:00				4pMU 1:30		5aMU 8:00

Acoustics ‘17 Boston

Photo courtesy of Greater Boston Convention and Visitors Bureau

25–29 June 2017

Joint Meeting of the

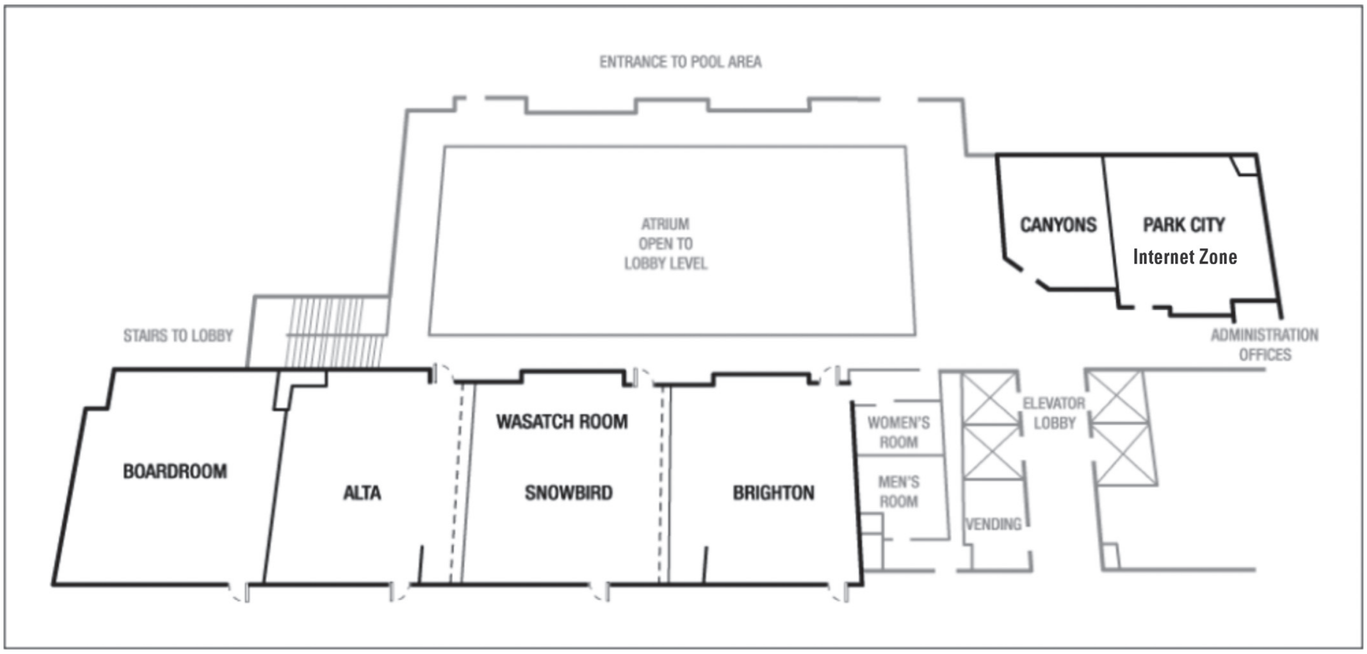
Acoustical Society of America

European Acoustics Association

The Acoustical Society of America (ASA) and the European Acoustics Association (EAA) invite acousticians from around the world to participate in **Acoustics ’17 Boston** to be held 25-29 June 2017 in Boston, Massachusetts, USA. A broad range of topics in acoustics will be covered in technical sessions and keynote lectures. Presentations on emerging topics are especially encouraged. Social events, student events, and an accompanying persons program will be organized. The best features of meetings of both organizations will be combined to offer a premier venue for presenting your work to an international audience.

Boston is the capitol and largest city in Massachusetts and is one of the oldest cities in the United States. It is on the Atlantic coast and is home to many historic sites dating back to the American Revolution, in addition to many other cultural and recreational features. The climate in June is very pleasant and ideal for arranging visits before and after the meeting.

Please join us!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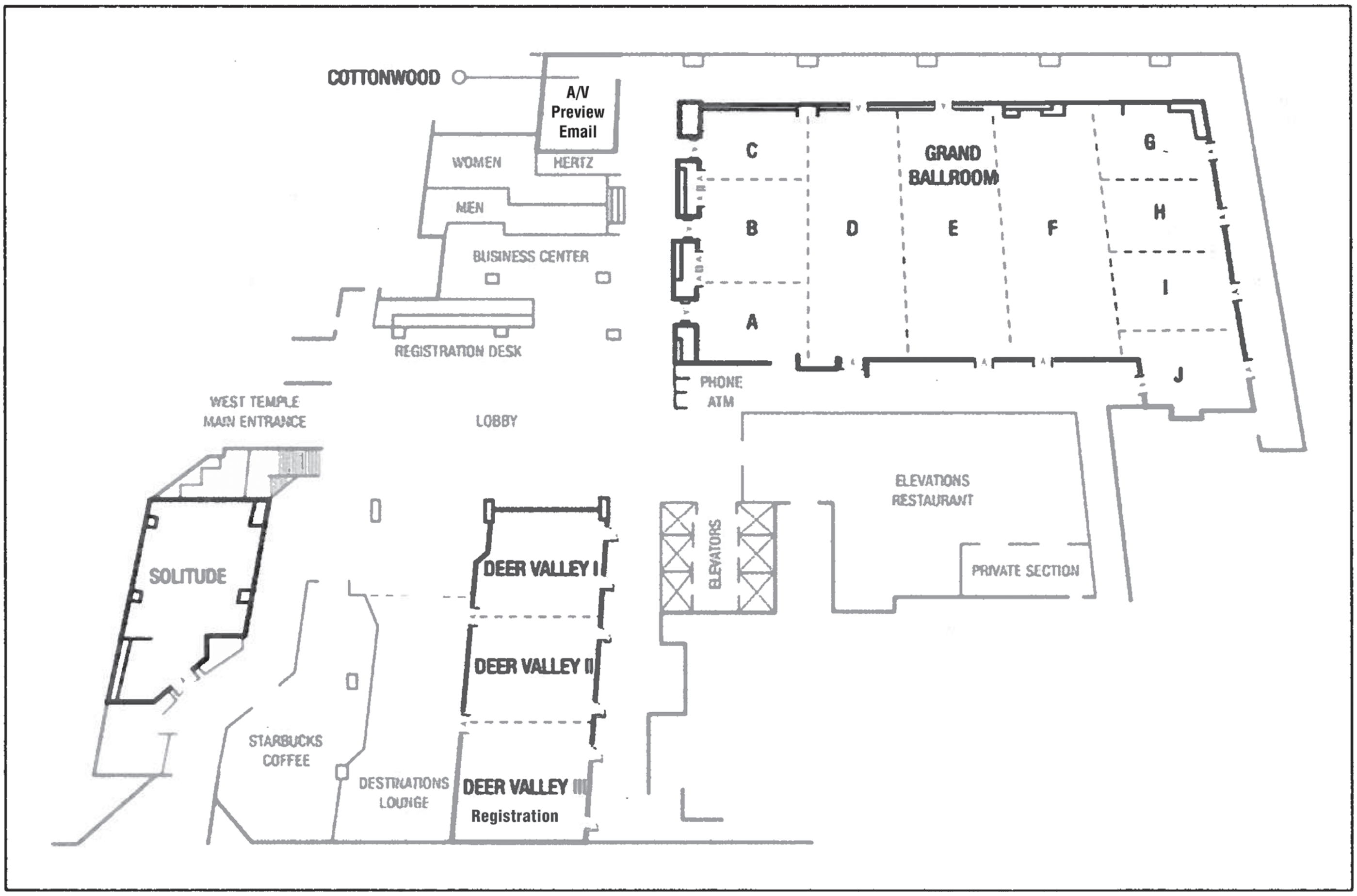
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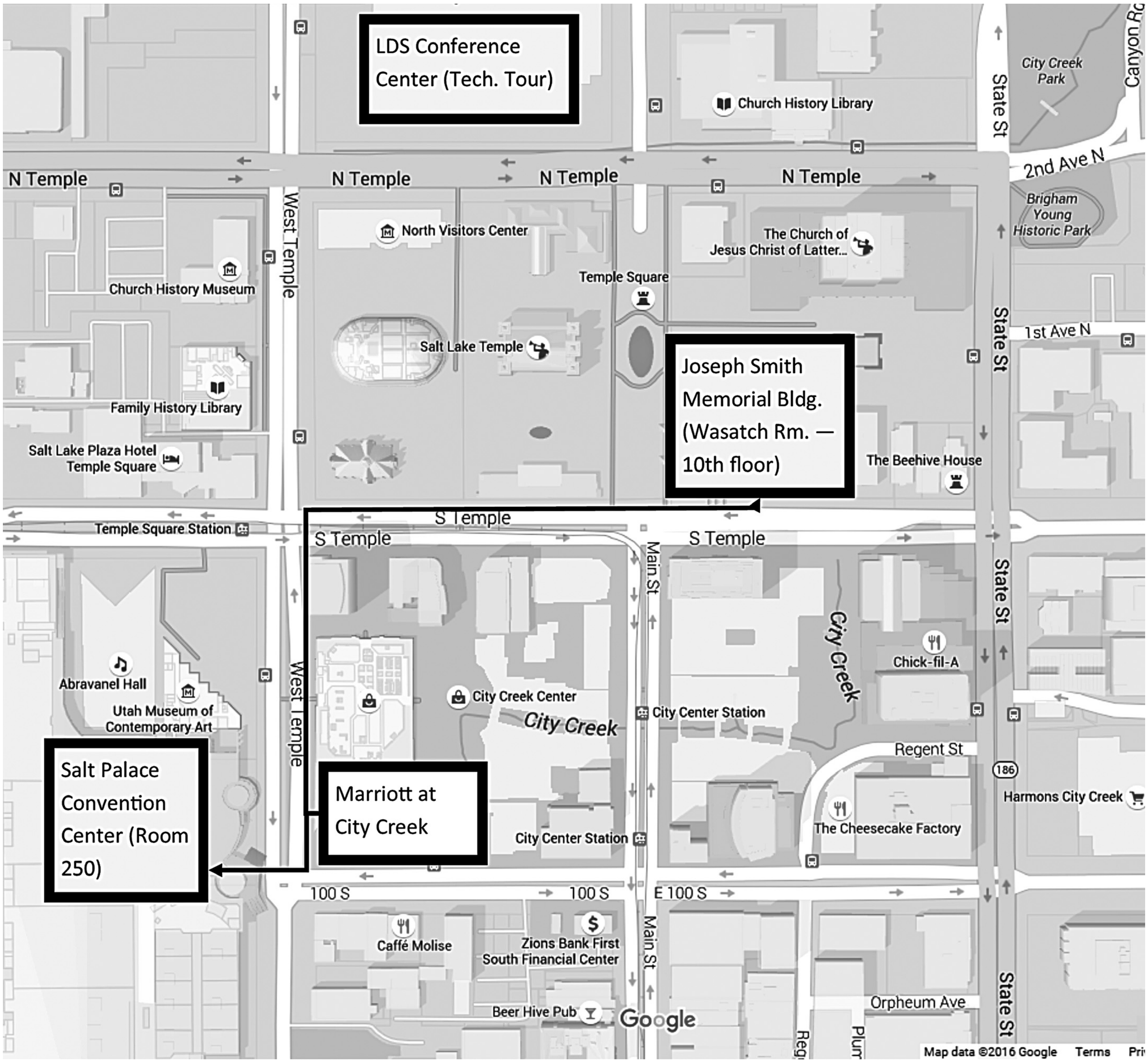
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TECHNICAL PROGRAM CALENDAR
171st Meeting
Salt Lake City, Utah
23–27 May 2016

MONDAY MORNING

9:00 1aAA **Architectural Acoustics, Noise, and ASA Committee on Standards:** Classroom Acoustics. Salon I

9:00 1aEDa **Education in Acoustics:** Hands-On Acoustics Demonstrations for Middle- and High-School Students. Salon D

10:00 1aEDb **Education in Acoustics, Public Relations, and Student Council:** Strategies for Effectively Communicating Science to Policy Makers, the Media, and the Public. Salon A

8:45 1aNS **Noise and ASA Committee on Standards:** Community Noise I. Salon B/C

8:15 1aPA **Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics:** Computational Methods in Physical Acoustics I. Salon H

8:00 1aPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics Potpourri (Poster Session). Salon E/F

MONDAY AFTERNOON

1:00 1pAA **Architectural Acoustics and Speech Communication:** Sound System Design, Optimization, and Intelligibility. Salon E

1:30 1pAB **Animal Bioacoustics and Psychological and Physiological Acoustics:** Comparative Hearing–Honoring Dick Fay. Salon G

5:30 1pED **Education in Acoustics and Women in Acoustics:** Listen Up and Get Involved. Salon D

2:15 1pID **Interdisciplinary and Student Council:** Introductions to Technical Committees. Salon A

1:30 1pNS **Noise and ASA Committee on Standards:** Community Noise II. Salon B/C

1:30 1pPA **Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics:** Computational Methods in Physical Acoustics II. Salon H

1:00 1pSAa **Structural Acoustics and Vibration and Musical Acoustics:** Analysis of Vibration Based Musical Instruments. Salon J

2:40 1pSAb **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration. Salon J

1:00 1pSC **Speech Communication:** Acoustics and Perception of Speech (Poster Session). Salon F

1:30 1pSP **Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics:** Small Unmanned Aerial Vehicle (UAV) Detection, Tracking, and Classification. Salon I

MONDAY EVENING

7:00 1eID **Interdisciplinary:** Tutorial Lecture on Acoustic Metamaterials: From Theory to Practice. Salon B/C

TUESDAY MORNING

8:00 2aAAa **Architectural Acoustics and Noise:** Forensic Studies from Noise Identification to Solutions in the Built Environment. Salon E

8:30 2aAAb **Architectural Acoustics and National Council of Acoustical Consultants:** Student Design Competition. Salon F

8:00 2aAO **Acoustical Oceanography and Signal Processing in Acoustics:** Acoustic Consistency of Ocean Models. Salon I

8:00 2aBAa **Biomedical Acoustics:** Acoustic Radiation Force and Elastography. Snowbird/Brighton

10:00 2aBAb **Biomedical Acoustics:** Therapeutic Ultrasound and Microbubbles. Snowbird/Brighton

8:00 2aEA **Engineering Acoustics:** General Topics in Engineering Acoustics I. Salon A

10:25 2aED **Education in Acoustics:** Education in Acoustics General Topics. Salon I

8:00 2aMU **Musical Acoustics:** Voice Registration in Amplified and Unamplified Singing. Salon B/C

8:00 2aNSa **Noise, Signal Processing in Acoustics, Architectural Acoustics, and ASA Committee on Standards:** Noise Measurements With Mobile Apps. Salon G

10:40 2aNSb **Noise, Structural Acoustics and Vibration, and Architectural Acoustics:** Noise and Vibration Impacts from Crossfit Training Facilities. Salon G

8:00 2aPA **Physical Acoustics and Biomedical Acoustics:** Vortex Beams and Radiation Torque Physics I. Salon H

8:00 2aPP **Psychological and Physiological Acoustics and Signal Processing in Acoustics:** Approaches to Improve Speech Understanding in Noise. Salon D

8:00	2aSC	**Speech Communication:** Intelligibility, Hearing Impairment, Aging (Poster Session). Salon F
8:00	2aSP	**Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics:** Acoustic Array Systems and Signal Processing I. Solitude
8:30	2aUW	**Underwater Acoustics:** Target Physics and Scattering. Salon J

TUESDAY AFTERNOON

1:00	2pAA	**Architectural Acoustics, Noise, and Engineering Acoustics:** Predicting and Controlling Heating, Ventilating, and Air Conditioning Systems Noise. Snowbird/ Brighton
1:00	2pABa	**Animal Bioacoustics and Underwater Acoustics:** Cetacean Bioacoustics. Salon I
3:30	2pABb	**Animal Bioacoustics:** Animal Bioacoustics Poster Session. Salon I
1:00	2pBA	**Biomedical Acoustics:** Biomedical Acoustics Student Paper Competition (Poster Session). Salon F
1:10	2pEA	**Engineering Acoustics:** General Topics in Engineering Acoustics II. Salon A
1:00	2pMU	**Musical Acoustics:** Pitch, Dynamics, and Vowel Tuning in Choral Voice. Salon B/C
1:30	2pNS	**Noise and Signal Processing in Acoustics:** Statistical Learning Techniques in Noise Research. Salon G
1:00	2pPA	**Physical Acoustics and Biomedical Acoustics:** Vortex Beams and Radiation Torque Physics II. Salon H
1:00	2pPPa	**Psychological and Physiological Acoustics:** Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical and Physiological Collaborations. Salon D
4:25	2pPPb	**Psychological and Physiological Acoustics:** Auditory Neuroscience Prize Lecture. Salon H
1:00	2pSAa	**Structural Acoustics and Vibration:** Real-World Instructive Case Studies in Structural Acoustics and Vibration. Salon E
3:40	2pSAb	**Structural Acoustics and Vibration and Signal Processing in Acoustics:** Wavenumber Transform Methods. Salon B/C
2:10	2pSC	**Speech Communication, Psychological and Physiological Acoustics, Architectural Acoustics, and ASA Committee on Standards:** Intelligibility Challenges: Speakers, Listeners, and Situations. Salon J
1:00	2pSP	**Signal Processing in Acoustics:** Comparison of Beamforming, Matched-Field Processing, and Time Reversal Techniques. Solitude

WEDNESDAY MORNING

8:30	3aAA	**Architectural Acoustics:** Relating Perception to Room Acoustics Measurements and Metrics in Performing Arts Venues I. Salon B/C
8:15	3aAB	**Animal Bioacoustics, Noise, and ASA Committee on Standards:** Effects of Noise on Animals. Salon I
8:15	3aBA	**Biomedical Acoustics:** Controlled Drug Delivery and Release with Focused Ultrasound. Salon H
8:00	3aED	**Education in Acoustics:** Education in Acoustics: Student Posters. Salon G
9:00	3aMU	**Musical Acoustics and Education in Acoustics:** Teaching Musical Acoustics Courses and Laboratories at Any Level. Solitude
8:30	3aPA	**Physical Acoustics:** Atmospheric Acoustic Phenomena I. Salon J
7:55	3aPP	**Psychological and Physiological Acoustics:** Quantitative Methodology in Both Physiological and Psychophysical Data Analysis Workshop. Salon D
8:30	3aSA	**Structural Acoustics and Vibration:** Computational Methods in Structural Acoustics and Vibration. Snowbird/Brighton
8:00	3aSC	**Speech Communication and Psychological and Physiological Acoustics:** Gender Effects in Speech Production and Perception. Salon F
8:30	3aSP	**Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics:** Acoustic Array Systems and Signal Processing II. Salon E
8:00	3aUW	**Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Sediment Characterization Using Direct and Inverse Techniques I. Salon A

WEDNESDAY AFTERNOON

1:15	3pAA	**Architectural Acoustics:** Relating Perception to Room Acoustics Measurements and Metrics in Performing Arts Venues II. Salon B/C
1:30	3pAB	**Animal Bioacoustics:** Airborne/Automatic Animal Bioacoustics. Salon I
1:00	3pBA	**Biomedical Acoustics:** Focused Ultrasound for Brain Treatments. Salon H
1:30	3pID	**Interdisciplinary:** Hot Topics in Acoustics. Salon D
1:15	3pPA	**Physical Acoustics:** Atmospheric Acoustic Phenomena II. Salon J
1:00	3pPP	**Psychological and Physiological Acoustics:** Beyond the Audiogram:

Influence of Supra-Threshold Deficits. Snowbird/Brighton

1:00 3pSC **Speech Communication:** Variation and Gender Effects (Poster Session). Salon G

1:30 3pUW **Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Sediment Characterization Using Direct and Inverse Techniques II. Salon A

THURSDAY MORNING

8:30 4aAA **Architectural Acoustics:** Emerging Parametric/Generative Design Tools in Architectural Acoustics. Salon G

8:00 4aAO **Acoustical Oceanography and Animal Bioacoustics:** Noise Impacts from the Industrialization of the Outer Continental Shelf and High Seas. Salon I

8:00 4aNS **Noise, ASA Committee on Standards, Animal Bioacoustics, Engineering Acoustics, and Physical Acoustics:** Wind Turbine Noise I. Salon B/C

8:00 4aPA **Physical Acoustics:** Multiple Scattering I. Salon H

8:00 4aPP **Psychological and Physiological Acoustics:** Temporal Aspects of Auditory Processing (Poster Session). Salon F

8:30 4aSAa **Structural Acoustics and Vibration, Architectural Acoustics, Noise, and ASA Committee on Standards:** Building Isolation from Seismic and Groundborne Vibration. Salon J

10:50 4aSAb **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Nuclear-Powered Thermoacoustics. Salon J

8:00 4aSC **Speech Communication:** Non-Native Speech Perception and Production (Poster Session). Salon E

10:30 4aSP **Signal Processing in Acoustics and Underwater Acoustics:** Detection and Estimation in Uncertain Acoustic Environments I. Snowbird/Brighton

8:00 4aUW **Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Sediment Characterization Using Direct and Inverse Techniques III. Salon A

THURSDAY AFTERNOON

1:00 4pAA **Architectural Acoustics and Musical Acoustics:** Opera Rehearsal and Performance Spaces. Salon E

1:00 4pAO **Acoustical Oceanography and Animal Bioacoustics:** Acoustical Oceanographic Tools for the Study of Marine Ecosystems. Salon I

1:30 4pBA **Biomedical Acoustics:** Imaging. Salon G

1:30 4pMU **Musical Acoustics:** Session in Honor of William J. Strong. Solitude

1:30 4pNS **Noise, ASA Committee on Standards, Animal Bioacoustics, Engineering Acoustics, and Physical Acoustics:** Wind Turbine Noise II. Salon B/C

1:00 4pPAa **Physical Acoustics:** General Topics in Physical Acoustics I. Salon F

1:00 4pPAb **Physical Acoustics:** Multiple Scattering II. Salon H

1:30 4pPP **Psychological and Physiological Acoustics and Speech Communication:** Lessons from Interrupted Speech: Methods and Models. Salon J

1:30 4pSC **Speech Communication and Signal Processing in Acoustics:** Constraints, Strategies, and Economies of Effort in Speech Production: Joseph S. Perkell's Contribution to Speech Science. Salon D

1:30 4pSP **Signal Processing in Acoustics and Underwater Acoustics:** Detection and Estimation in Uncertain Acoustic Environments II. Snowbird/Brighton

1:30 4pUW **Underwater Acoustics:** Acoustic Propagation in the Ocean. Salon A

FRIDAY MORNING

8:00 5aAA **Architectural Acoustics:** A Variety of Interesting Research and Observations in Architectural Acoustics. Salon I

8:00 5aMU **Musical Acoustics:** General Topics in Musical Acoustics. Solitude

8:30 5aNS **Noise:** Topics in Noise Control. Salon A

8:00 5aPA **Physical Acoustics:** General Topics in Physical Acoustics II. Salon H

7:55 5aPP **Psychological and Physiological Acoustics:** Spatial Hearing. Salon D

8:00 5aSCa **Speech Communication:** Phonetics of Under-Documented Languages I. Salon B/C

10:05 5aSCb **Speech Communication:** Phonetics of Under-Documented Languages II (Poster Session). Salon F

9:00 5aSCc **Speech Communication:** Speech Production (Poster Session). Salon E

8:00 5aSP **Signal Processing in Acoustics:** General Topics in Signal Processing. Salon G

8:30 5aUW **Underwater Acoustics:** Underwater Noise. Salon J

SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Mon, 23 May, 7:30 a.m.	Executive Council	Snowbird/Brighton
Mon, 23 May, 3:30 p.m.	Technical Council	Snowbird/Brighton
Mon, 23 May, 6:30 p.m.	Technical Council dinner	Alta
Tue, 24 May, 7:00 a.m.	ASA Press Editorial Board & Books+	250 A/B Salt Palace
Tue, 24 May, 7:30 a.m.	Panel on Public Policy	Boardroom
Tue, 24 May, 11:30 a.m.	Audit	Room 1532
Tue, 24 May, 11:30 a.m.	Student Council	Deer Valley 1
Tue, 24 May, 11:45 a.m.	Editorial Board	250 A/B, Salt Palace
Tue, 24 May, 12:30 p.m.	Prizes & Special Fellowships	Boardroom
Tue, 24 May, 1:30 p.m.	Meetings	Deer Valley 1
Tue, 24 May, 4:30 p.m.	Newman Fund Advisory	Boardroom
Tue, 24 May, 5:00 p.m.	Women in Acoustics	250 A/B, Salt Palace
Wed, 25 May, 7:00 a.m.	International Research & Education	Boardroom
Wed, 25 May, 7:00 a.m.	College of Fellows	Room 1532
Wed, 25 May, 7:00 a.m.	POMA, JASA-EL, Publication Policy	250 A, Salt Palace
Wed, 25 May, 7:00 a.m.	Regional and Student Chapters	250 B, Salt Palace
Wed, 25 May, 11:00 a.m.	Medals and Awards	Boardroom
Wed, 25 May, 11:30 a.m.	Public Relations	Alta
Wed, 25 May, 12:00 noon	Membership	Room 1532
Wed, 25 May, 1:30 p.m.	AS Foundation Board	Alta
Wed, 25 May, 5:15 p.m.	Education in Acoustics	Salon I
Wed, 25 May, 5:30 p.m.	Acoustics Today Editorial Board	Boardroom
Thu, 26 May, 7:00 a.m.	Archives & History	Deer Valley 1
Thu, 26 May, 7:30 a.m.	Tutorials	Alta
Thu, 26 May, 7:30 a.m.	Investments	Boardroom
Thu, 26 May, 2:00 p.m.	Publishing Services	Boardroom
Thu, 26 May, 2:00 p.m.	Strategic Plan Champions	Alta
Thu, 26 May, 4:30 p.m.	External Affairs	Alta
Thu, 26 May, 4:30 p.m.	Internal Affairs	Deer Valley 1
Fri, 27 May, 7:00 a.m.	Technical Council	Snowbird/Brighton
Fri, 27 May, 11:00 a.m.	Executive Council	Snowbird/Brighton

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 24 May, 4:30 p.m.	Engineering Acoustics	Salon A
Tue, 24 May, 7:30 p.m.	Acoustical Oceanography	Salon J
Tue, 24 May, 7:30 p.m.	Animal Bioacoustics	Salon I
Tue, 24 May, 7:30 p.m.	Architectural Acoustics	Snowbird/Brighton
Tue, 24 May, 7:30 p.m.	Physical Acoustics	Salon H
Tue, 24 May, 7:30 p.m.	Psychological and Physiological Acoustics	Salon B/C
Tue, 24 May, 7:30 p.m.	Structural Acoustics and Vibration	Salon G
Wed, 25 May, 7:30 p.m.	Biomedical Acoustics	Salon H
Wed, 26 May, 7:30 p.m.	Musical Acoustics	Salon B/C
Wed, 25 May, 7:30 p.m.	Signal Processing in Acoustics	Salon G
Thu, 26 May, 7:30 p.m.	Noise	Salon D
Thu, 26 May, 7:30 p.m.	Speech Communication	Salon G
Thu, 26 May, 7:30 p.m.	Underwater Acoustics	Salon J

STANDARDS COMMITTEES AND WORKING GROUPS

Mon, 23 May, 2:30 p.m.	S12/WG56-Noise in Parks	Boardroom
Mon, 23 May, 5:00 p.m.	S2-Mechanical Vibration and Shock	Room 1532
Mon, 23 May, 7:00 p.m.	ASACOS Steering	Boardroom
Tue, 24 May, 7:30 a.m.	ASACOS	Alta
Tue, 24 May, 9:15 a.m.	Standards Plenary Group	Alta
Tue, 24 May, 11:00 a.m.	S12-Noise	Alta
Tue, 24 May, 2:00 p.m.	S3/SC1-Animal Bioacoustics	Alta
Tue, 24 May, 3:15 p.m.	S3-Bioacoustics	Alta
Tue, 24 May, 4:45 p.m.	S1-Acoustics	Alta
Wed, 25 May, 8:45 a.m.	S12/WG15-Environmental Noise	Deer Valley 1
Wed, 25 May, 9:30 a.m.	S12/WG18-Criteria for Room Noise	Alta
Thu, 26 May, 6:30 p.m.	S3/SC1/WG6-Auditory Evoked Potential Testing	Boardroom

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon-Thu, 23-26 May 7:30 a.m. - 5:00 p.m. Fri, 27 May 7:30 a.m. - 12:00 noon	Registration	Deer Valley
Mon-Thu, 23-26 May, 7:00 a.m. - 5:00 p.m. Fri, 27 May, 7:00 a.m. - 12:00 noon	E-mail	Cottonwood
Mon-Thu, 23-26 May, 7:00 a.m. - 5:00 p.m. Fri, 6 Nov 7:00 a.m. - 12:00 noon	Internet Zone	Park City
Mon-Thu, 23-26 May 7:00 a.m.- 5:00 p.m. Fri, 27 May 7:00 a.m. - 12:00 noon	A/V Preview	Cottonwood
Mon-Fri, 23-27 May 8:00 a.m. - 10:00 a.m.	Accompanying Persons	Room 1531
Mon-Thu, 23-26 May, 8:00 a.m. - 5:00 p.m.	Gallery of Acoustics	Grand Ballroom Foyer
Sun, 22 May, 1:00 p.m. - 5:00 p.m. Mon, 23 May 8:30 a.m. - 12:30 p.m.	Short Course	Solitude
Mon-Fri, 23-27 May 9:45 a.m. - 10:30 a.m.	Coffee Break	Ballroom prefunction
Tue-Thu, 24-26 May 12:00 p.m. - 1:30 p.m.	Resume Help Desk	Grand Ballroom Foyer
Mon, 23 May, 2:45 p.m. - 5:15 p.m.	Technical Tour-LDS Conference	Marriott Lobby Center
Mon, 23 May 5:00 p.m. - 5:30 p.m.	New Student Orientation	Salon A
Mon, 2 Nov 5:30 p.m. - 6:45 p.m.	Student Meet and Greet	Salon I/J
Tue, 24 May, 6:00 p.m. - 7:30 p.m.	Social Hour	Salon D/E/F
Wed, 25 May, 11:45 a.m. - 1:30 p.m.	Women in Acoustics Luncheon	250 A/B Salt Palace
Wed, 25 May, 3:30 p.m.	Annual Membership Meeting	Salon E/F
Wed, 25 May, 3:30 p.m. - 5:30 p.m.	Plenary Session/Awards Ceremony	Salon E/F
Wed, 25 May, 6:00 p.m. - 8:00 p.m.	Student Reception	250 A/B Salt Palace
Wed, 25 May, 6:00 p.m.	Brigham Young University Dinner	Joseph Smith Memorial Bldg., Empire Room
Wed, 25 May, 8:00 p.m. - 12:00 midnight	ASA Jam	Salon E/F
Thu, 26 May, 11:45 a.m. - 1:30 p.m.	Society Luncheon and Lecture	Joseph Smith Memorial
Thu, 26 May, 6:00 p.m. - 7:30 p.m.	Social Hour	Salon E/F
Fri, 27 May 3:30 p.m. - 8:30 p.m. Sat, 28 May 8:00 a.m. - 12:30 p.m.	Early Career Retreat	Salon E

171st Meeting of the Acoustical Society of America

The 171st meeting of the Acoustical Society of America will be held Monday through Friday, 23–27 May 2016 at the Salt Lake Marriott Downtown at City Creek Hotel, Salt Lake City, Utah, USA.

SECTION HEADINGS

1. HOTEL INFORMATION
2. TRANSPORTATION AND TRAVEL DIRECTIONS
3. STUDENT TRANSPORTATION SUBSIDIES
4. MESSAGES FOR ATTENDEES
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6. ASSISTIVE LISTENING DEVICES
7. TECHNICAL SESSIONS
8. TECHNICAL SESSION DESIGNATIONS
9. HOT TOPICS SESSION
10. WILLIAM AND CHRISTINE HARTMANN PRIZE IN AUDITORY NEUROSCIENCE AND THE AUDITORY NEUROSCIENCE PRIZE LECTURE
11. TUTORIAL LECTURE
12. SHORT COURSE
13. STUDENT DESIGN COMPETITION
14. GALLERY OF ACOUSTICS
15. RESUME HELP DESK
16. TECHNICAL COMMITTEE OPEN MEETINGS
17. TECHNICAL TOUR
18. ANNUAL MEMBERSHIP MEETING
19. PLENARY SESSION AND AWARDS CEREMONY
20. ANSI STANDARDS COMMITTEES
21. COFFEE BREAKS
22. A/V PREVIEW ROOM
23. PROCEEDINGS OF MEETINGS ON ACOUSTICS
24. E-MAIL ACCESS AND INTERNET ZONE
25. SOCIALS
26. SOCIETY LUNCHEON AND LECTURE
27. STUDENTS MEET MEMBERS FOR LUNCH
28. STUDENT EVENTS: NEW STUDENTS ORIENTATION, MEET AND GREET, STUDENT RECEPTION
29. WOMEN IN ACOUSTICS LUNCHEON
30. JAM SESSION
31. BRIGHAM YOUNG UNIVERSITY ACOUSTICS RECEPTION AND DINNER
32. ACCOMPANYING PERSONS PROGRAM
33. WEATHER
34. TECHNICAL PROGRAM ORGANIZING COMMITTEE
35. MEETING ORGANIZING COMMITTEE
36. PHOTOGRAPHING AND RECORDING
37. ABSTRACT ERRATA
38. GUIDELINES FOR ORAL PRESENTATIONS
39. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS
40. GUIDELINES FOR USE OF COMPUTER PROJECTION
41. DATES OF FUTURE ASA MEETINGS

1. HOTEL INFORMATION

The Salt Lake Marriott Downtown at City Creek Hotel is the headquarters hotel where most meeting events will be held.

The cut-off date for reserving rooms at special rates has passed. Please contact the Salt Lake Marriott Downtown at City Creek Hotel for lodging availability (75 South West Temple, Salt Lake City, UT 84101, T: (801) 531-0800.

2. TRANSPORTATION AND TRAVEL DIRECTIONS

Salt Lake City is served by many major airlines through the Salt Lake City International Airport which is located 6 miles from downtown Salt Lake City.

Transportation from the Salt Lake City International Airport to the Salt Lake Marriott Downtown at City Creek Hotel and other downtown Salt Lake City hotels:

Information. The Information Booth is located in the baggage claim area in each terminal's main common area.

Light Rail Service. The TRAX light rail system provides transportation from the airport to downtown Salt Lake City, with a TRAX stop located less than one block from the Salt Lake City Marriott Downtown at City Creek Hotel. The train stop is located at the south end of Terminal One. A train leaves the Airport for the city center every 15 minutes on weekdays and every 20 minutes on weekends, with a 12 minute transit time. The current one-way fare is USD $2.50.

Major car rental companies. Nearly every major car rental company is represented at Salt Lake City International Airport. Rental car counters are located just outside the terminal buildings in the Short Term Parking Structure.

Airport Shuttle shared-ride service. There are shuttle companies departing the Salt Lake City International Airport for downtown hotels. Express Shuttle is the largest service, and offers shuttle service for USD $8.00 (one way). They can be found at the Ground Transportation Desk located in the baggage claim area. Phone 801-596-1600 or 800-397-0773.

Taxicabs and limousines. Taxis are available outside the terminal at Salt Lake City International Airport. The meeting hotel is 6 miles from the airport (approximately 10 minutes), with fares averaging USD $25.00 one way. All fares are metered.

3. STUDENT TRANSPORTATION SUBSIDIES

To encourage student participation, limited funds are available to defray partially the cost of travel expenses of students to attend Acoustical Society meetings. Instructions for applying for travel subsidies are given in the Call for Papers which can be found online at http://acousticalsociety.org. The deadline for the present meeting has passed but this information may be useful in the future.

4. MESSAGES FOR ATTENDEES

A message board will be available in the registration area for posting messages for attendees.

5. REGISTRATION

Registration is required for all attendees and accompanying persons. Registration badges must be worn in order to participate in technical sessions and other meeting activities. Registration will open on Monday, 23 May, at 7:30 a.m. in the Deer Valley Room (see floor plan on page A12).

Checks or travelers checks in U.S. funds drawn on U.S. banks and Visa, MasterCard and American Express credit cards will be accepted for payment of registration. Meeting attendees who have pre-registered may pick up their badges and registration materials at the pre-registration desk.

The registration fees (in USD) are $570 for members of the Acoustical Society of America; $645 for non-members, $150 for Emeritus members (Emeritus status pre-approved by ASA), $295 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $90 for ASA Student members, $135 for students who are not members of ASA, $0 for Nonmember Undergraduate Students (proof of undergraduate status required), and $150 for accompanying persons.

One-day registration is available at $295 for members and $360 for nonmembers (one-day means attending the meeting on only one day either to present a paper and/or to attend sessions). A nonmember who pays the $645 nonmember registration fee and simultaneously applies for Associate Membership in the Acoustical Society of America will be given a $50 discount off their dues payment for 2016 dues.

Invited speakers who are members of the Acoustical Society of America are expected to pay the registration fee, but nonmember invited speakers who participate in the meeting only on the day of their presentation may register without charge. The registration fee for nonmember invited speakers who wish to participate for more than one day is $115 and includes a one-year Associate Membership in the ASA upon completion of an application form.

Special note to students who pre-registered online: You will also be required to show your student id card when picking-up your registration materials at the meeting.

6. ASSISTIVE LISTENING DEVICES

The ASA has purchased assistive listening devices (ALDs) for the benefit of meeting attendees who need them at technical sessions. Any attendee who will require an assistive listening device should advise the Society in advance of the meeting by writing to: Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; asa@acousticalsociety.org

7. TECHNICAL SESSIONS

The technical program includes 96 sessions with 933 papers scheduled for presentation during the meeting.

A floor plan of the Marriott Hotel appears on pages A11 and A12. Session Chairs have been instructed to adhere strictly to the printed time schedule, both to be fair to all speakers and to permit attendees to schedule moving from one session to another to hear specific papers. If an author is not present to deliver a lecture-style paper, the Session Chairs have been instructed either to call for additional discussion of papers already given or to declare a short recess so that subsequent papers are not given ahead of the designated times.

Several sessions are scheduled in poster format, with the display times indicated in the program schedule.

8. TECHNICAL SESSION DESIGNATIONS

The first character is a number indicating the day the session will be held, as follows:

1-Monday, 23 May
2-Tuesday, 24 May
3-Wednesday, 25 May
4-Thursday, 26 May
5-Friday, 27 May

The second character is a lower case “a” for a.m., “p” for p.m., or “e” for evening corresponding to the time of day the session will take place. The third and fourth characters are capital letters indicating the primary Technical Committee that organized the session using the following abbreviations or codes:

AA Architectural Acoustics
AB Animal Bioacoustics
AO Acoustical Oceanography
BA Biomedical Acoustics
EA Engineering Acoustics
ED Education in Acoustics
ID Interdisciplinary
MU Musical Acoustics
NS Noise
PA Physical Acoustics
PP Psychological and Physiological Acoustics
SA Structural Acoustics and Vibration
SC Speech Communication
SP Signal Processing in Acoustics
UW Underwater Acoustics

In sessions where the same group is the primary organizer of more than one session scheduled in the same morning or afternoon, a fifth character, either lower-case “a,” “b,” or “c” is used to distinguish the sessions. Each paper within a session is identified by a paper number following the session-designating characters, in conventional manner. As hypothetical examples: paper 2pEA3 would be the third paper in a session on Tuesday afternoon organized by the Engineering Acoustics Technical Committee; 3pSAb5 would be the fifth paper in the second of two sessions on Wednesday afternoon sponsored by the Structural Acoustics and Vibration Technical Committee.

Note that technical sessions are listed both in the calendar and the body of the program in the numerical and alphabetical order of the session designations rather than the order of their starting times. For example, session 3aAA would be listed ahead of session 3aAO even if the latter session began earlier in the same morning.

9. HOT TOPICS SESSION

A Hot Topics session, 3pID, will be held on Wednesday, 25 May, at 1:30 p.m. in Salon D. Papers will be presented on current topics in the fields of Architectural Acoustics, Psychological and Physiological Acoustics, Musical Acoustics and Education in Acoustics.

10. WILLIAM AND CHRISTINE HARTMANN PRIZE LECTURE IN AUDITORY NEUROSCIENCE AND THE AUDITORY NEUROSCIENCE PRIZE LECTURE

The 2016 William and Christine Hartmann Prize in Auditory Neuroscience will be awarded to Alan R. Palmer of the MRC Institute of Hearing Research, UK, at the Plenary Session on Wednesday, 25 May. Alan Palmer will present the Auditory Neuroscience Prize Lecture titled "Bridging the chasm: Animal physiology and human psychophysics" on Tuesday, 24 May, at 4:25 p.m. in Session 2pPPb in Salon H.

11. TUTORIAL LECTURE

A tutorial presentation on "Acoustic Metamaterials: From Theory to Practice" will be given by Andrew N. Norris, Professor at Rutgers University, on Monday, 23 May at 7:00 p.m.in Salon B/C. The tutorial will provide a comprehensive introduction to ideas such as acoustic anisotropy, allowing sound to have speed that depends on direction, and transformation acoustics which is the basis for acoustic cloaking. Lecture notes will be available at the meeting in limited supply. To partially defray the cost of the lecture, a registration fee is charged. The fee is USD $25. Students with current ID cards is USD $7.

12. SHORT COURSE

A short course titled Bayesian Data Analysis will be given in two parts: Sunday, 22 May, from 1:00 p.m. to 5:00 p.m. and Monday, 23 May, from 8:30 a.m. to 12:30 p.m. in the Solitude Room.

Statistical modeling is central to essentially all quantitative fields of research. Theoretical and computational developments in recent years have made Bayesian statistical methods widely available. Bayesian methods provide powerful, flexible tools for analyzing and drawing inferences from complex (and simple) data sets. This course will cover (1) data processing and visualization using the statistical programming language R and (2) the basics of Bayesian data analysis using R, JAGS, and Stan.

The course instructor is Noah Silbert, Assistant Professor in the Department of Communication Sciences and Disorders at the University of Cincinnati. He has an MA in ESL from the University of Hawaii, an MS in Applied Statistics and a PhD in Linguistics and Cognitive Science from Indiana University. The full registration fee is USD $300 (USD $125 for students) and covers attendance, instructional materials and coffee breaks. The number of attendees is limited to 30.

13. STUDENT DESIGN COMPETITION

The 2016 Student Design Competition will be held on Tuesday, 24 May, in session 2aAAb at 8:30 a.m. in Salon F. This competition is intended to encourage students in the disciplines of architecture, engineering, physics, and other curricula that involve building design and/or acoustics to express their knowledge of architectural acoustics and noise control in the design of a facility in which acoustical considerations are of significant importance. The Student Design Competition is sponsored by the ASA Technical Committee on Architectural Acoustics, with support from the Wenger Foundation, the Robert Bradford Newman Student Award Fund, and the National Council of Acoustical Consultants.

14. GALLERY OF ACOUSTICS

The Technical Committee on Signal Processing in Acoustics will sponsor the 15th Gallery of Acoustics at the meeting. Its purpose is to enhance ASA meetings by providing a setting for researchers to display their work to all meeting attendees in a forum emphasizing the diversity, interdisciplinary, and artistic nature of acoustics. The Gallery at Salt Lake City will be unique relative to previous Galleries, in that only images (posters, pictures, etc.) will be accepted.

The Gallery will be held in the Grand Ballroom Foyer Monday through Thursday, 23–26 May, 8:00 a.m. to 5:00 p.m. A cash prize of USD $400 and USD $200 will be awarded to the winning and first runner-up entries, respectively. Meeting attendees are asked to rank-order the entries using the ballot distributed with registration materials.

15. RESUME HELP DESK

Are you interested in applying for graduate school, a postdoctoral opportunity, a research scientist position, a faculty opening, or other position involving acoustics? If you are, please stop by the ASA Resume Help Desk in the Deer Valley Foyer near the registration desk. Members of the ASA experienced in hiring will be available to look at your c/v, cover letter, and research & teaching statements to provide tips and suggestions to help you most effectively present yourself in today's competitive job market. The ASA Resume Help Desk will be staffed on Tuesday, Wednesday, and Thursday during the lunch hour (12 noon – 1:30 p.m.) for walk-up meetings. Appointments during these three lunch hours will also be available via a sign-up sheet posted in the registration area.

16. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday at the Marriott Hotel. The schedule and rooms for each Committee meeting are given on page A18.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussions.

17. TECHNICAL TOUR

A tour of the LDS Conference Center in downtown Salt Lake City will be conducted, Monday, 23 May 2016. A group will leave from the Marriott Hotel lobby at 2:45 pm. The Conference Center is a leisurely 5-10 minute walk from the hotel. The tour will summarize many of the acoustic and audiovisual challenges presented by a 21,000 seat auditorium used for everything from spoken word to organ/choral music, and will summarize many lessons learned in the 15 years since opening. The tour will include an organ demonstration/recital, as well as a demonstration of the audiovisual system, including the LARES electronic reverberation system. Attendees will then have the chance to go inside the organ loft as well as the catwalks above the stage.

The cost is USD $5 and registration will be limited to 40 attendees.

18. ANNUAL MEMBERSHIP MEETING

The Annual Membership Meeting of the Acoustical Society of America will be held at 3:30 p.m. on Wednesday, 25 May 2016, in Salon E/F at the Salt Lake Marriott Downtown at City Creek Hotel, 75 South West Temple, Salt Lake City, UT 84101.

19. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 25 May, at 3:30 p.m. in Salon E/F.

The William and Christine Hartmann Prize in Auditory Neuroscience will be presented to Alan R. Palmer.

The Distinguished Service Citation will be presented to Susan B. Blaeser, the R. Bruce Lindsay Award will be presented to Megan S. Ballard, the Helmholtz-Rayleigh Interdisciplinary Silver Medal will be presented to Armen Sarvazyan, and the Gold Medal will be presented to Whitlow W. L. Au.

Certificates will be presented to Fellows elected at the Jacksonville meeting of the Society. See page 2127 for a list of fellows.

20. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will be held at the Salt Lake City meeting on the days and times listed in the Schedule of Committee Meetings and Other Events on page A18.

Meetings of selected advisory working groups are often held in conjunction with Society meetings and are listed in the Schedule or on the standards bulletin board in the registration area, e.g., S12/WGI8-Room Criteria.

People interested in attending and in becoming involved in working group activities should contact the ASA Standards Manager for further information about these groups, or about the ASA Standards Program in general, at the following address: Susan Blaeser, ASA Standards Manager, Standards Secretariat, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; T.: 631-390-0215; F: 631-923-2875; E: asastds@acousticalsociety.org

21. COFFEE BREAKS

Morning coffee breaks will be held each day from 9:45 a.m. to 10:30 a.m. in the Ballroom Prefunction Foyer.

22. A/V PREVIEW ROOM

The Cottonwood Room will be set up as an A/V preview room for authors' convenience, and will be available on Monday through Thursday from 7:00 a.m. to 5:00 p.m. and Friday from 7:00 a.m. to 12:00 noon.

23. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Salt Lake City meeting will have a published proceedings, and submission is optional. This is an open access journal, so that its articles are available in pdf format without charge to anyone in the world for downloading. Authors who are scheduled to present papers at the meeting are encouraged to prepare a suitable version in pdf format that will appear in POMA. Further information regarding POMA can be found at the site http://scitation.aip.org/content/asa/journal/poma. Published papers from previous meeting can be seen at the site http://asadl/poma.

24. E-MAIL ACCESS AND INTERNET ZONE

Computers providing e-mail access will be available 7:00 a.m. to 5:00 p.m., Monday to Thursday and 7:00 a.m. to 12:00 noon on Friday in the Cottonwood Room. The Internet Zone will be available 7:00 a.m. to 5:00 p.m. Monday to Thursday and 7:00 a.m. to 12:00 noon on Friday in the Park City Room.

Wifi will be available in all ASA meeting rooms and spaces.

25. SOCIALS

Complimentary buffet socials with cash bar will be held on Tuesday and Thursday evenings from 6:00 p.m. to 7:30 p.m. in Salon D/E/F at the Marriott Hotel.

All meeting attendees are invited to attend the social hours. The ASA hosts these social hours to provide a relaxing setting for meeting attendees to meet and mingle with their friends and colleagues as well as an opportunity for new members and first-time attendees to meet and introduce themselves to others in the field. A second goal of the socials is to provide a sufficient meal so that meeting attendees can attend the Technical Committees meetings that begin immediately after the socials.

26. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture will be held on Thursday, 21 May, at 11:45 a.m. in the Wasatch Room on the 10th floor of the Joseph Smith Memorial Building which is a 6-minute walk from the Marriott. The luncheon is open to all attendees and their guests. The speaker is Professor Jeremy Grimshaw, Associate Dean in the College of Fine Arts and Communications at Brigham Young University and the director of Gamelan Bintang Wahyu. He will present a lecture and Balinese music demonstration entitled, "Exactly Out of Tune: Acoustics, Aesthetics, Culture, and Ritual in Balinese Gamelan Music.

Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 25 May. The cost is USD $30.00 per ticket.

27. STUDENTS MEET MEMBERS FOR LUNCH

The ASA Education Committee arranges one-on-one lunch meetings between students and ASA members. The purpose is to make it easier for students to meet and interact with members at Acoustical Society meetings. Each lunch pairing is arranged separately. Students who are interested should contact Dr. David Blackstock, University of Texas at Austin, by email dtb@austin.utexas.edu. Please provide your name, university, department, degree you are seeking (BS, MS, or PhD), research field, acoustical interests, your supervisor's name, days you are free for lunch, and abstract number (or title) of any paper(s) you are presenting. The sign-up deadline is 12 days before the start of the meeting, but an earlier sign-up is strongly encouraged. Each participant pays for his/her own meal.

28. STUDENT EVENTS: NEW STUDENTS ORIENTATION, MEET AND GREET, STUDENT RECEPTION

Follow the student twitter throughout the meeting @ASAStudents.

A New Students Orientation will be held from 5:00 p.m. to 5:30 p.m. on Monday, 23 May, in Salon A for all students to learn about the activities and opportunities available for students at the Salt Lake City meeting. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in Salon I/J. Refreshments and a cash bar will be available. Students are encouraged to attend the tutorial lecture on Metamaterials which begins at 7:00 p.m. in Salon B/C. The student fee for the tutorial is USD $12.

The Students' Reception will be held on Wednesday, 25 May, from 6:00 p.m. to 8:00 p.m. in Room 250 A/B at the Salt Palace Convention Center which is across the street from the Marriott.

This reception, sponsored by the Acoustical Society of America and supported by the National Council of Acoustical Consultants, will provide an opportunity for students to meet informally with fellow students and other members of the Acoustical Society. All students are encouraged to attend, especially students who are first time attendees or those from smaller universities.

Students will find a sticker to place on their name tags identifying them as students in their registration envelopes. Although wearing the sticker is not mandatory, it will allow for easier networking between students and other meeting attendees.

Students are encouraged to refer to the student guide, also found in their envelopes, for important program and meeting information pertaining only to students attending the ASA meeting.

They are also encouraged to visit the official ASA Student Home Page at http://asastudentcouncil.org/ to learn more about student involvement in ASA.

29. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:45 a.m. on Wednesday, 25 May, in Room 250 A/B at the Salt Palace Convention Center which is across the street from the Marriott. Meeting participants who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 25 May. The fee is USD $30 for non-students and USD $15 for students.

30. JAM SESSION

You are invited to Salon E/F on Wednesday night, 25 May, from 8:00 p.m. to midnight for the JAM SESSION. Bring your axe, horn, sticks, voice, or anything else that makes music. Musicians and non-musicians are all welcome to attend. A full PA system, backline equipment, guitars, bass, keyboard, and drum set will be provided. All attendees will enjoy live music, a cash bar with snacks, and all-around good times. Don't miss out.

31. BRIGHAM YOUNG UNIVERSITY ACOUSTICS RECEPTION AND DINNER

Between 1907–1915, the first three graduates in Physics at Brigham Young University were Harvey Fletcher, Carl Eyring, and Vern Knudsen – all of whom figure prominently in the ASA's rich history. To mark this centennial and to celebrate several decades of its strong tradition in acoustics, the BYU Acoustics Research Group is hosting a dinner for alumni and friends on the evening of Wednesday, 25 May, at 6:00 p.m. in the Empire Room at the Joseph Smith Memorial Building which is a 6-minute walk from the Marriott.

Current students, alumni, faculty and friends of the BYU Acoustics Research Group, and their accompanying persons, are welcome to attend. Additional information about the event is available at http://acoustics.byu.edu. Tickets cost USD $30 each and must be purchased by Tuesday, 24 May.

32. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Salt Lake City meeting. The registration fee is USD $125 for pre-registration by 25 April and USD $150 at the meeting.

A hospitality room for accompanying persons will be open at the City Creek Marriott from 8:00 a.m. to 10:00 a.m., Monday through Friday. A brief overview of surrounding sites will be presented on Monday at 9:00 a.m. On Tuesday, there will be a morning Trolley tour of the downtown Salt Lake City area for USD $30/person. There will also be a day tour to the Winter Olympic park, Park City Mountain Resort, Deer Valley, and Downtown Park City (http://saltlakecityguidedtours.com/parkcitytour.htm) on Wednesday that will include stops at the outlet malls for USD $60/person. Brochures of local attractions and dining options will be provided.

33. WEATHER

The weather is generally very pleasant in the latter part of May, but cool weather can occur. For the week of the ASA meeting, temperatures are typically in the mid 70s during the day, dropping into the lower 50s at night.

34. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Brian E. Anderson, Jonathan Blotter, Cochairs; David P. Knobles, Acoustical Oceanography; Benjamin Taft, Animal Bioacoustics; Damian J. Doria, Ian B. Hoffman, Architectural Acoustics; Siddhartha Sikdar, Biomedical Acoustics; Andrew A. Piacsek, Eon King, Education in Acoustics; Kenneth M. Walsh, Engineering Acoustics; Andrew C.H. Morrison, Musical Acoustics; Eric L. Reuter, James E. Phillips, Noise; Michael R. Haberman, Physical Acoustics; Christopher A. Brown, Psychological and Physiological Acoustics; Ning Xiang, Said Assous, Signal Processing in Acoustics; Alexander L. Francis, Melissa Baese-Berk, Tuuli Morrill, Speech Communication; Benjamin M. Shafer, Structural Acoustics and Vibration; Anthony Bonomo, Todd Hefner, Underwater Acoustics; Matthew Kamrath, Brent Reichman, Student Council.

35. MEETING ORGANIZING COMMITTEE

Scott D. Sommerfeldt, Kent L. Gee, Meeting Cochairs; Jonathan Blotter, Brian E. Anderson, Technical Program Cochairs; Sarah Rollins, Audio Visual; Tracianne B. Neilsen, Food/Beverage; Sarah H. Ferguson, Signs; Timothy E. Doyle, Student Volunteers; Cole V. Duke, Technical Tour; Samuel J. Anderson, ASA School; Timothy W. Leishman, Society Luncheon & Lecture; Skyler G. Jennings, Internet Zone.

36. PHOTOGRAPHING AND RECORDING

Photographing and recording during regular sessions are not permitted without prior explicit permission of the presenter.

37. ABSTRACT ERRATA

This meeting program is Part 2 of the April 2016 issue of *The Journal of the Acoustical Society of America*. Corrections, for printer's errors only, may be submitted for publication in the Errata section of the *Journal*.

38. GUIDELINES FOR ORAL PRESENTATIONS

Preparation of Visual Aids

See the enclosed guidelines for computer projection.

- Allow at least one minute of your talk for each Powerpoint slide. No more than 12 slides for a 15-minute talk (with 3 minutes for questions and answers).
- Minimize the number of lines of text on one visual aid. 12 lines of text should be a maximum. Include no more than 2 graphs/plots/figures on a single slide. Generally, too little information is better than too much.
- Presentations should contain simple, legible text that is readable from the back of the room.
- Characters should be at least 0.25 inches (6.5 mm) in height to be legible when projected. A good rule of thumb is that text should be 20 point or larger (including labels in inserted graphics). Anything smaller is difficult to read.
- Make symbols at least 1/3 the height of a capital letter.
- For computer presentations, use all of the available screen area using landscape orientation with very thin margins. If your institution's logo must be included, place it at the bottom of the slide.
- Sans serif fonts (e.g., Arial, Calibri, and Helvetica) are much easier to read than serif fonts (e.g., Times New Roman) especially from afar. Avoid thin fonts (e.g., the horizontal bar of an e may be lost at low resolution thereby registering as a c.)
- Do not use underlining to emphasize text. It makes the text harder to read.
- All axes on figures should be labeled and the text size for labels and axis numbers or letters should be large enough to read.
- No more than 3–5 major points per slide.
- Consistency across slides is desirable. Use the same background, font, font size, etc. across all slides.
- Use appropriate colors. Avoid complicated backgrounds and do not exceed four colors per slide. Backgrounds that change from dark to light and back again are difficult to read. Keep it simple.
- If using a dark background (dark blue works best), use white or yellow lettering. If you are preparing slides that may be printed to paper, a dark background is not appropriate.
- If using light backgrounds (white, off-white), use dark blue, dark brown or black lettering.
- DVDs should be in standard format.

Presentation

- Organize your talk with introduction, body, and summary or conclusion. Include only ideas, results, and concepts that can be explained adequately in the allotted time. Four elements to include are:
  (1) Statement of research problem
  (2) Research methodology
  (3) Review of results
  (4) Conclusions
- Generally, no more than 3–5 key points can be covered adequately in a 15-minute talk so keep it concise.
- Rehearse your talk so you can confidently deliver it in the allotted time. Session Chairs have been instructed to adhere to the time schedule and to stop your presentation if you run over.
- An A/V preview room will be available for viewing computer presentations before your session starts. It is advisable to preview your presentation because in most cases you will be asked to load your presentation onto a computer, which may have different software or a different configuration from your own computer.
- Arrive early enough so that you can meet the session chair, load your presentation on the computer provided, and familiarize yourself with the microphone, computer slide controls, laser pointer, and other equipment that you will use during your presentation. There will be many presenters loading their materials just prior to the session so it is very important that you check that all multi-media elements (e.g., sounds or videos) play accurately prior to the day of your session.
- Each time you display a visual aid the audience needs time to interpret it. Describe the abscissa, ordinate, units, and the legend for each figure. If the shape of a curve or some other feature is important, tell the audience what they should observe to grasp the point. They won't have time to figure it out for themselves. A popular myth is that a technical audience requires a lot of technical details. Less can be more.
- Turn off your cell phone prior to your talk and put it away from your body. Cell phones can interfere with the speakers and the wireless microphone.

39. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS

Content

- The poster should be centered around two or three key points supported by the title, figures, and text.
- The poster should be able to "stand alone." That is, it should be understandable even when you are not present to explain, discuss, and answer questions. This quality is highly desirable since you may not be present the entire time posters are on display, and when you are engaged in discussion with one person, others may want to study the poster without interrupting an ongoing dialogue.
- To meet the "stand alone" criteria, it is suggested that the poster include the following elements, as appropriate:
- Background
- Objective, purpose, or goal
- Hypotheses
- Methodology
- Results (including data, figures, or tables)
- Discussion
- Implications and future research
- References and Acknowledgment

Design and Layout

- A board approximately 8 ft. wide × 4 ft. high will be provided for the display of each poster. Supplies will be available for attaching the poster to the display board. Each board will be marked with an abstract number.

- Typically posters are arranged from left to right and top to bottom. Numbering sections or placing arrows between sections can help guide the viewer through the poster.
- Centered at the top of the poster, include a section with the abstract number, paper title, and author names and affiliations. An institutional logo may be added. Keep the design relatively simple and uncluttered. Avoid glossy paper.

Lettering and text
- Font size for the title should be large (e.g., 70-point font)
- Font size for the main elements should be large enough to facilitate readability from 2 yards away (e.g., 32 point font). The font size for other elements, such as references, may be smaller (e.g., 20–24 point font).
- Sans serif fonts (e.g., Arial, Calibri, Helvetica) are much easier to read than serif fonts (e.g., Times New Roman).
- Text should be brief and presented in a bullet-point list as much as possible. Long paragraphs are difficult to read in a poster presentation setting.

Visuals
- Graphs, photographs, and schematics should be large enough to see from 2 yards (e.g., 8 × 10 inches).
- Figure captions or bulleted annotation of major findings next to figures are essential. To ensure that all visual elements are "stand alone," axes should be labeled and all symbols should be explained.
- Tables should be used sparingly and presented in a simplified format.

Presentation
- Prepare a brief oral summary of your poster and short answers to likely questions in advance.
- The presentation should cover the key points of the poster so that the audience can understand the main findings. Further details of the work should be left for discussion after the initial poster presentation.
- It is recommended that authors practice their poster presentation in front of colleagues before the meeting. Authors should request feedback about the oral presentation as well as poster content and layout.

Other suggestions
- You may wish to prepare reduced-size copies of the poster (e.g., 8 1/2 × 11 inch sheets) to distribute to interested audience members.

40. GUIDELINES FOR USE OF COMPUTER PROJECTION

- A PC computer with audio playback capability and a projector will be provided in each meeting room on which all authors who plan to use computer projection should load their presentations.
- Authors should bring computer presentations on a USB drive to load onto the provided computer and should arrive at the meeting rooms at least 30 minutes before the start of their sessions.
- Assistance in loading presentations onto the computers will be provided.
- Note that only PC format will be supported so authors using Macs to prepare their presentation must save their presentations so that the projection works when the presentation is run from the PC in the session room. Also, authors who plan to play audio or video clips during their presentations should insure that their sound (or other) files are also saved on the USB drive and are also uploaded to the PC in the session room. Presenters should also check that the links to the sound (and other) files in the presentation still work after everything has been loaded onto the session room computer.

Using your own computer (only if you really need to!)
It is essential that each speaker who plans to use his/her own laptop connect to the computer projection system in the A/V preview room prior to session start time to verify that the presentation will work properly. Technical assistance is available in the A/V preview room at the meeting, but not in session rooms. Presenters whose computers fail to project for any reason will not be granted extra time.

General Guidelines
- Set your computer's screen resolution to 1024x768 pixels or to the resolution indicated by the AV technical support. If it looks OK, it will probably look OK to your audience during your presentation.
- Remember that graphics can be animated or quickly toggled among several options: Comparisons between figures may be made temporally rather than spatially.
- Animations often run more slowly on laptops connected to computer video projectors than when not so connected. Test the effectiveness of your animations before your assigned presentation time on a similar projection system (e.g., in the A/V preview room). Avoid real-time calculations in favor of pre-calculation and saving of images.
- If you will use your own laptop instead of the computer provided, connect your laptop to the projector during the question/answer period of the previous speaker. It is good protocol to initiate your slide show (e.g., run PowerPoint) immediately once connected, so the audience doesn't have to wait. If there are any problems, the session chair will endeavor to assist you, but it is your responsibility to ensure that the technical details have been worked out ahead of time.
- During the presentation have your laptop running with main power instead of using battery power to insure that the laptop is running at full CPU speed. This will also guarantee that your laptop does not run out of power during your presentation.

SPECIFIC HARDWARE CONFIGURATIONS

Macintosh
- Older Macs require a special adapter to connect the video output port to the standard 15-pin male DIN connector. Make sure you have one with you.
- Hook everything up before powering anything on. (Connect the computer to the RGB input on the projector).
- Turn the projector on and boot up the Macintosh. If this doesn't work immediately, you should make sure that your monitor resolution is set to 1024x768 for an XGA projector or at least 640x480 for an older VGA projector. (1024x768 will most always work.). You should also make sure that

your monitor controls are set to mirroring. If it's an older powerbook, it may not have video mirroring, but something called simulscan, which is essentially the same.

- Depending upon the vintage of your Mac, you may have to reboot once it is connected to the computer projector or switcher. Hint: you can reboot while connected to the computer projector in the A/V preview room in advance of your presentation, then put your computer to sleep. Macs thus booted will retain the memory of this connection when awakened from sleep.
- Depending upon the vintage of your system software, you may find that the default video mode is a side-by-side configuration of monitor windows (the test for this will be that you see no menus or cursor on your desktop; the cursor will slide from the projected image onto your laptop's screen as it is moved). Go to Control Panels, Monitors, configuration, and drag the larger window onto the smaller one. This produces a mirror-image of the projected image on your laptop's screen.
- Also depending upon your system software, either the Control Panels will automatically detect the video projector's resolution and frame rate, or you will have to set it manually. If it is not set at a commensurable resolution, the projector may not show an image. Experiment ahead of time with resolution and color depth settings in the A/V preview room (please don't waste valuable time adjusting the Control Panel settings during your allotted session time).

PC

- Make sure your computer has the standard female 15-pin DE-15 video output connector. Some computers require an adaptor.
- Once your computer is physically connected, you will need to toggle the video display on. Most PCS use either ALT-F5 or F6, as indicated by a little video monitor icon on the appropriate key. Some systems require more elaborate keystroke combinations to activate this feature. Verify your laptop's compatibility with the projector in the A/V preview room. Likewise, you may have to set your laptop's resolution and color depth via the monitor's Control Panel to match that of the projector, which settings you should verify prior to your session.

Linux

- Most Linux laptops have a function key marked CRT/LCD or two symbols representing computer versus projector. Often that key toggles on and off the VGA output of the computer, but in some cases, doing so will cause the computer to crash. One fix for this is to boot up the BIOS and look for a field marked CRT/LCD (or similar). This field can be set to Both, in which case the signal to the laptop is always presented to the VGA output jack on the back of the computer. Once connected to a computer projector, the signal will appear automatically, without toggling the function key. Once you get it working, don't touch it and it should continue to work, even after reboot.

41. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, or to obtain instructions for the preparation and submission of meeting abstracts, contact the Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; Telephone: 516-576-2360; Fax: 631-923-2875; E-mail: asa@acousticalsociety.org: Web: AcousticalSociety.org

172nd Meeting, Honolulu, Hawaii, 28 November – 2 December 2016

173rd Meeting, Acoustics '17 Boston, Boston, MA, 25–29 June 2017, Joint meeting of the Acoustical Society of America and the European Acoustics Association

174th Meeting, New Orleans, Louisiana, 4–8 December 2017

175th Meeting, Minneapolis, Minnesota, 7–11 May 2018

176th Meeting, Victoria, Canada, 5–9 November 2018

177th Meeting, Louisville, Kentucky, 13–17 May 2019

FIFTY YEAR AWARDS

"Gold" certificates in recognition of continuing interest in membership in the Society for half a century will be sent to the following members:

Yoichi Ando
Yosuhiko Arai
James W. Beauchamp
Charles F. Boyle
William S. Brown, Jr.
Robert D. Bruce
Courtney B. Burroughs
Robert C. Coffeen
H. Steven Colburn
Daniel E. Commins
Rufus L. Cook
Henry Cox
Lawrence A. Crum
George D. Curtis
Darshan S. Dosanjh
Floyd W. Emanuel
Lawrence L. Feth
Graeme E. Harding
Robert D. Hellweg, Jr.
Bruce D. Keller
Martin Klein
Moises Levy
Jerome E. Manning
Halcyon E. Morris
Mack J. Preslar
Roelof J. Ritsma
Aaron E. Rosenberg
Jack A. Shooter
James A. Simmons
Robert A. Strachan
David Strelioff
Maurice M. Taylor
Constantine Trahiotis
Gunther E. Urbanek
Istvan L. Ver
Alfred E. Victor
Teruji Yamamoto

TWENTY-FIVE YEAR AWARDS

The following individuals have been members of the Society continuously for twenty-five years. They will be sent to the following members:

Michael A. Ainslie
Daniel H. Ashmead
Douglas H. Bell
William E. Biker
Ocke-Schwen Bohn
Vance F. Breshears, Jr.
John R. Buck
Sean K. Bui
Alexander L. Butler
Dani M. Byrd
Antoine J. Chaigne
Shun-Hsyung Chang
Weinhwa Chiang
Grant B. Deane
Dimitri M. Donskoy
David R. Dowling
Ward R. Drennan
Brian D. Dushaw
Martin Ebel
David A. Eddins
Gayle M. Ayers-Elam
Donna Erickson
Mark A. Ericson
James B. Ewalt
John B. Fahnline
Michael J. Ferragamo
James J. Finneran
Peter A. Fryer
Philippe Guy
Bjorn Hagerman
Robert Hagiwara
Dan C. Halling
Helen M. Hanson
Mark A. Hasegawa-Johnson
Kathy J. Hays-Stang
Eric Healy
Laurie M. Heller
Avraham Hirschberg
Dehua Huang
John G. Hurt
Sven M. Ivansson
Sergio Jesus
Deborah A. Jue
Ahmet Kahraman
Kenneth H. Kaliski
Steven G. Kargl
Debra M. Kenney
Kilsoo S. Kim
Jay M. Kirsch
Mendel Kleiner
Robert D. Kollars
Jeffrey L. Krolik
Hidemi Kubota
Chang-ho Lee
John Loverde
Anthony P. Lyons
Mark J. McKenna
Gary S. Madaras
Edward R. Maniet, Jr.
Michael D. Max
Edward C. Monahan
Thomas W. Morse
Stefan Muenkner
Rochelle S. Newman
Man Wa Lawrence Ng
Keiji Oguchi
Kaoru Okabe
Toshiyuki T. O. Okano
Robert D. O'Neal
Scott D. Pfeiffer
Jorge L. Pind
James C. Preisig
Brigitte A. Quetglas
Henning Reetz
Bernard E. Richardson
Catherine L. Rogers
Jose Romero-Faus
Alan T. Rosen
Robert T. Sataloff
Michael V. Scanlon
Mark E. Schaffer
Anna Marie Schmidt
Lisa A. Schott
Franck C. Sgard
William P. Shofner
Julius O. Smith, III
Hee Chun Song
Pamela E. Souza
Brad H. Story
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Session 1aAA

Architectural Acoustics, Noise, and ASA Committee on Standards: Classroom Acoustics

Shiu-Keung Tang, Cochair
Department of Building Services Engineering, The Hong Kong Polytechnic University, Hong Kong, Hong Kong

Siu Kit Lau, Cochair
Dept. of Architecture, National Univ. of Singapore, Block SDE3, #01-06, 4 Architecture Drive, Singapore 117566, Singapore

Chair's Introduction—9:00

Invited Papers

9:05

1aAA1. Comparison of occupied and unoccupied noise levels in K-12 classrooms. Laura C. Brill and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St, Omaha, NE 68182-0816, lbrill@huskers.unl.edu)

Current research at the University of Nebraska—Lincoln aims to establish how indoor environmental conditions in K-12 school buildings impact student scholastic achievement. A large-scale *in-situ* survey is being undertaken to gather data on (1) indoor environmental conditions under different seasons or outdoor conditions (fall, winter, and spring), and (2) student's standardized test outcomes and demographics, as classroom aggregates with no individually identifiable information. In this paper, acoustic data gathered to date from 110 classrooms are mined to determine how occupied and unoccupied noise levels correlate. The ANSI Classroom Acoustics Standard S12.60 (2010) specifies guidelines for unoccupied background noise levels, but students learn in occupied spaces. Logged acoustic data over multiple school days are analyzed to determine if designing for unoccupied noise levels is appropriate to achieve desired occupied acoustic conditions. [Work supported by the United States Environmental Protection Agency Grant Number R835633.]

9:25

1aAA2. Cantonese speech intelligibility tests in two Hong Kong primary school classrooms. Shiu-Keung Tang (Dept. of Bldg. Services Eng., Hong Kong Polytechnic Univ., Hong Kong, Hong Kong, shiu-keung.tang@polyu.edu.hk) and Siu-Kit Lau (Dept. of Architecture, National Univ. of Singapore, Singapore, Singapore)

Speech transmission is a very important issue especially in early childhood education during which the children learn languages and pronunciations. There are evidences that the acoustical performance of a classroom can have significant impact on the learning progress and effectiveness of the children. The effect is not only observed in language subjects, but also in the numeracy related subjects. In this study, on-site speech intelligibility tests are conducted using newly developed phonetically balanced (in Cantonese) Chinese character lists suitable for the Year 5 Hong Kong primary school children (10–11 years old) in two classrooms. A trained speaker is recruited to speak out the characters. The classroom acoustical properties are also measured, but in the absence of the pupils because of statutory regulation. The sound of each Cantonese character in general consists of an initial consonant, a vowel, a final consonant, and a tone. A preliminary observation is that there is higher probability for the pupils to give wrong answers when the sound of a character lasts for relatively long duration, especially when there are abrupt temporal changes in the time-frequency spectra. Correct answers are usually given when the corresponding time-frequency spectral energy is more concentrated within a short duration.

9:45

1aAA3. Effect of room acoustics on speech perception by children with hearing loss. Z. Ellen Peng, Florian Pausch, and Janina Fels (Medical Acoust. Group, Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 95757, Germany, zpe@akustik.rwth-aachen.de)

To study speech perception of children with hearing loss in virtual acoustic environments (VAE), a pair of research hearing aids has been previously integrated in a real-time dynamic binaural reproduction system. The auralization included simulations of room acoustics using individualized head-related and hearing aid-related transfer functions (HRTF and HARTF). In this study, a release from masking paradigm by Cameron and Dillon (2007) was adopted in German to investigate speech intelligibility by children fitted with hearing aids under realistic classroom acoustics. When immersed in VAE, each child was asked to repeat sentences spoken by a target talker always located at 0° azimuth, while two distractor talkers were continuously telling unfamiliar Grimm stories. The speech reception threshold (SRT) at 50% intelligibility was measured adaptively by changing the target talker speech level. A total of eight conditions were tested with each child by changing spatial cues (target-distractor collocated versus spatially separated at 90° horizontally), pitch cues (target-distractor sharing the same versus different voice), and room acoustics (0.4 versus 1.2 s mid-frequency averaged reverberation time). Measured SRTs from children with hearing loss will be compared with those from age-matched normal-hearing children. [Work supported by EU Seventh Framework- iCARE, Improving Children's Auditory Rehabilitation.]

10:05–10:20 Break

10:20

1aAA4. Optimizing the signal to noise ratio in classrooms using passive acoustics. Peter D'Antonio (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

Adults with normal hearing require a roughly 0 dB signal-to-noise ratio (SNR) for good speech intelligibility in classrooms and lecture halls. However, values in excess of 15 dB are needed to compensate for neurological immaturity, sensorineural and conductive hearing losses, language proficiency, and excessive reverberation. ANSI 12.60 addresses ways to lower the noise interference due to background levels and reverberation time. However, simply reducing the mid-high frequency reverberation time is not sufficient and may lead to masking of the high frequency consonant information, needed for speech intelligibility, with lower frequency vowel sounds. Research has shown that increasing the signal by including early reflections is also needed. This presentation will discuss a classroom design, along with several potential acoustical treatments, which aims to passively increase the SNR and improve speech intelligibility. These acoustic tools include a multi-layer, broad bandwidth absorptive panel, a hybrid diffusive/absorptive panel, an absorptive wood system simultaneously offering early reflections and mid-low frequency sound absorption and an LED lighting fixture system that simultaneously provides light and sound diffusion, as well as variable color and intensity for task lighting that has been shown to improve reading speed, increase test scores, and reduce hyperactivity.

10:40

1aAA5. A case of classroom acoustic treatment in Tsinghua University. Xiang Yan and Hui Li (Acousic Lab of School of Architecture, Tsinghua Univ., Rm. 104, Main Bldg., Tsinghua University, Beijing, Beijing 100084, China, lihuisylvia@aliyun.com)

Room 114 is an 100 audience classroom located in the Architecture Department Hall of Tsinghua University with unsatisfactory acoustic effects. This paper briefly introduces the whole process of acoustic design for this classroom which included document survey, acoustic test, acoustic simulation analysis, scale model test, auralization simulation, etc. After weighing both the acoustic and visual effects of three designs, the executives determined the final one to put into effect. This case suggests that technology has been able to well solve the problem of the sound quality of the classroom in universities, but how to persuade managers adopt these technologies, maybe more difficult in classroom acoustics which need further studies.

11:00

1aAA6. A global index of acoustic assessment in elemental school classrooms. Jianxin Peng (School of Phys. and Optoelectronics, South China Univ. of Technol., Wushan Rd. No. 381, Guangzhou, Guangdong 10640, China, phjxpeng@163.com)

This paper describes the results of a study aimed at developing a method for optimizing the acoustic quality of a classroom. A global index of acoustic assessment in elemental school classrooms was developed for this purpose. It is especially important in classrooms, where suitable conditions should be provided to convey speech communication for teachers and students. The paper presents a method for assessing the acoustic quality of classrooms based on a single number global index and taking into account a number of factors affecting the outcome of the assessment, such as speech intelligibility, noise level, reverberation time, speech sound pressure level, and the age of students. A method is proposed based on an analysis of factors determining whether a room meets applicable acoustic requirements.

11:20

1aAA7. A field measurement study on the acoustic environment of primary and secondary schools in Southwestern China. Hui Xie and Yang He (School of Architecture & Urban Planning, Chongqing Univ., Chongqing, Chongqing 400045, China, yanshencun@hotmail.com)

A number of studies over the past 30 years have shown that the environmental and classroom noise have detrimental impacts on children's academic performance. This paper presents the field measurement results of the acoustic environment in 15 typical primary and secondary schools in Southwestern China. RT and three different background noise levels (outdoor, indoor with and without fan) were taken into account for all the 30 classrooms (two rooms per school). Compared with UK's schools, Chinese classrooms are normally bigger, with the average volume of 180 m^3. Both RT and background levels at most of the measured schools largely exceeded the suggested values, as guided by the relevant Chinese standard. RT ranged from 0.62 s to 2.63 s, whereas the maximum background noise levels were as high as 74 dBA. A possible reason might be the lack of appropriate acoustic treatments in those selected schools.

Contributed Paper

11:40

1aAA8. Speech accommodation to room acoustics: Reverberation time and clarity. Pasquale Bottalico (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Lansing, MI 48910, pb@msu.edu), Simone Graetzer, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Vocal effort is a physiological entity that accounts for changes in voice production as vocal loading increases, which can be quantified in terms of sound pressure level (SPL). This study investigates how vocal effort is affected by speaking style, room acoustics, and short-term vocal fatigue. Twenty subjects were recorded while reading a text at normal and loud volumes in anechoic, semi-reverberant, and reverberant rooms in the presence of babble noise. The acoustics in each environment were modified by increasing the strength of the early reflections in the talker position. The subjects answered questions addressing their perception of vocal effort, comfort, control, and the clarity of their own voice. SPL variation for each subject was measured per task. It was found that SPL and self-reported effort increased in the loud style and decreased when the reflective panels were present and when reverberation time increased. In contrast, self-reported comfort and control decreased in the loud style, while self-reported clarity increased when panels were present. SPL was higher for females than for males. The lowest magnitude of apparent short-term vocal fatigue was experienced by talkers in the semi-reverberant room. The results indicate that reflections may be used to reduce vocal effort without modifying reverberation time.

Session 1aEDa

Education in Acoustics: Hands-On Acoustics Demonstrations for Middle- and High-School Students

Kent L. Gee, Chair
Brigham Young University, N243 ESC, Provo, UT 84602

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomena. In this session "Hands-On" demonstrations will be set-up for a group of middle school students from the Salt Lake City area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students' educational development. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Any acousticians wanting to participate in this fun event should e-mail Kent Gee (kentgee@byu.edu).

MONDAY MORNING, 23 MAY 2016 SALON A, 10:00 A.M. TO 12:00 NOON

Session 1aEDb

Education in Acoustics, Public Relations, and Student Council: Strategies for Effectively Communicating Science to Policy Makers, the Media, and the Public

Andrew A. Piacsek, Cochair
Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

Kerri Seger, Cochair
Scripps Institution of Oceanography, 9331 Discovery Way, Apt. C, La Jolla, CA 92037

Chair's Introduction—10:00

Invited Papers

10:05

1aEDb1. Ten tips for communicating with Congress—and why it is worth doing. Rachel Carr (American Inst. of Physics/ASA, Washington, DC, carr.physics@gmail.com)

As this year's AIP-ASA Congressional Science Fellow, I have a up-close view of how scientists' voices are received on Capitol Hill. I am often surprised, both by what catches the attention of legislators and their staffs and by how impactful a brief meeting can be. With research funding decisions and acoustics-related public policy at stake, there can be major benefits to engaging with Congress and planning these interactions strategically. I will share some suggestions for doing that, based on my observations and advice from House and Senate staffers. While the general guidelines of science communication apply, I will focus on the timelines and tendencies of this unique institution and areas where acoustical scientists may be most connected.

10:25

1aEDb2. Beyond decibels: Inspiring informed noise management in U.S. National Parks. Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

Noise was not a significant threat to parks when the Organic Act was signed. As recently as five decades ago, road and aircraft traffic were less than a third of their present levels. In response to emerging noise issues, the NPS created an overflights program in 2000, which has evolved into the Natural Sounds and Night Skies Division. Noise is not inevitable. It is crucial to raise awareness of this growing threat without evoking a sense of defeat or resignation. Successful presentations start by arousing a need for action and hinting at the prospects for successful fulfilment. Compelling visualizations are the foundation of effective science presentations. Maps of noise exposure on continental and national park scales have been especially effective. For other scientific graphics, consider starting with an illustration of the simplest message, and sequentially adding elements to support the primary message. Often the most important decision is what to leave out of a figure. It is crucial to connect any measurement or metric to something that your audience cares about and make these units relateable. If you must use decibels, be sure to connect these to sensible effects.

10:45

1aEDb3. Turn on the lights!. Amy O'Donoghue (Deseret News, 5349 S. 6300 West, Hooper, UT 84111, amyjoi@deseretnews.com)

Over the decades that I've been a reporter and editor in the news business, I've never met a technical expert who was not just a bit pleased and more than a little passionate about the work they do once you coax it out of them. A crime scene technician relishes every little detail in the forensic evidence they collect. Meteorologists enthuse over El Nino or become giddy at the latest iteration of the Madden-Julian oscillation. Nuclear scientists are fascinated by radioactive daughter products. Despite this passion, most scientists don't make a splash about what they're up to. They go about work every day, picking apart technical details in obscurity, even acting mildly surprised if an outsider shows an interest. What cuts across many "subject matter" disciplines is curious modesty and often the mistaken notion by experts that their work is boring or of little public interest. My advice? Speak up. We're anxiously listening, and while you may have to repeat yourself a couple of times for us to "get it," we want to understand. We have the ability to shine a light on what you do, and explain why it is important, so help us help spread your important messages.

11:05

1aEDb4. Examples of both expected and unexpected interactions and outcomes when conveying science and technology to traditional print and modern media journalists. Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

It has been my experience with both traditional print journalists, and with modern media journalists, that unexpected interactions and outcomes sometimes occur. These can have both desirable and undesired consequences for the scientist. Some examples of these interactions will be given, which are associated with quite diverse technical topics, ranging from press and radio interviews covering work on the soundscape of a threatened species (*Ceratotherium simum simum*, the southern white rhinoceros), to a radio interview on the acoustics of coffee roasting that included a surprise guest, to an underwater photo shoot with a National Geographic photographer covering research on an underwater noise abatement system. Some guidelines will be presented that aim to minimize the undesired outcomes.

11:25–11:30 Break

11:30–12:00 Panel Discussion

MONDAY MORNING, 23 MAY 2016 SALON B/C, 8:45 A.M. TO 11:45 A.M.

Session 1aNS

Noise and ASA Committee on Standards: Community Noise I

Eric L. Reuter, Chair
Reuter Associates, LLC, 10 Vaughan Mall, Suite 201A, Portsmouth, NH 03801

Invited Papers

8:45

1aNS1. The city of the future—Noise initiative in the "Year of Science 2015". André Fiebig (Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de), Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Berlin, Germany), and Klaus Genuit (Acoust. GmbH, Herzogenrath, Germany)

In the context of the German Year of Science 2015, the issue noise in cities was addressed by means of a special initiative in order to raise noise awareness and to stimulate public participation. The "Sound of the City" initiative taking place within the framework of the Year of Science 2015 was organized and supported by the Federal Ministry of Education and Research (BMBF). By using different media, people were invited to report on how their cities sound. They were encouraged to upload audio recordings of their acoustical environments, to relate the sounds to the respective places and to describe the sounds and their perception. In general, the participants should consider sounds, which they typically associate with their cities. This action led to a comprehensive data base of audio files, which all have in common that the sounds bear a particular meaning for the participants. In a subsequent study, all submitted sounds were subject to a detailed analysis to investigate from different viewpoints the distinctive features of the sounds. The paper will present the results of the analyses and discuss the findings with respect to the potential of such public participation in the field of community noise.

9:05

1aNS2. Analysis of 500 noise ordinances. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

The noise ordinances for the 500 largest communities in the United States were collected and analyzed to determine the various techniques, metrics and criteria communities use to regulate noise (such as sound pressure level, nuisance, disturbing the peace, plainly audible, minimum distance, time of day, etc.). A survey of a random sample of those communities was conducted to determine the effectiveness of the noise regulation techniques, metrics, and criteria.

9:25

1aNS3. An abbreviated history and recent developments regarding community noise and soundscape management in New Orleans. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

New Orleans has one of the most complicated soundscapes of the cities of America, and with it a veritable gumbo of people and their opinions about sound and the city. Recent efforts to revise the ordinance and reduce sound levels from the Vieux Carre Entertainment District met with an encouraging response from residents and business owners alike, yet failed in City Council. The New Orleans Health Department has begun a program of proactive education of entertainment workers in the last year. This paper will touch on the origins of the city soundscape, early regulations, and then discuss in greater detail a summary of the recent developments in soundscape management.

9:45

1aNS4. Community noise annoyance: Connecting regression methodologies and theoretical models. D. K. Wilson (Cold Regions Res. Eng. Lab., U.S. Army Engineer Res. Dev. Ctr., U.S. Army Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755-1290, D. Keith.Wilson@usace.army.mil), Edward T. Nykaza (Construction Eng. Res. Lab., U.S. Army Engineer Res. Dev. Ctr., Champaign, IL), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), Nicole M. Wayant (Geospatial Res. Lab., U.S. Army Engineer Res. Dev. Ctr., Alexandria, VA), and Chandler M. Armstrong (Construction Eng. Res. Lab., U.S. Army Engineer Res. Dev. Ctr., Champaign, IL)

Recent statistical analyses indicate that the large, seemingly random scatter in community noise annoyance survey data results from systematic variations in noise tolerance between communities. This paper describes a simple hierarchical (or multilevel) modeling approach to community noise, which incorporates variations in tolerance among individuals, communities, and surveys. A regression methodology consistent with this hierarchical perspective, namely, that of partial pooling with a generalized linear model, is also described and applied to survey data for transportation noise. From this approach, it is shown that the "community tolerance level" (CTL) model proposed by Fidell *et al.* [J. Acoust. Soc. Am. **130**, 791–806 (2011)] is a particular case of partial pooling regression, in which it is implied that the annoyance responses of all surveyed communities are independent and grow at a fixed rate with increasing day-night level. Simulations and a meta-analysis of transportation noise annoyance studies provide strong evidence that individual- and community-level variations have distinct statistical signatures, and that the growth of annoyance is not fixed between communities.

10:05–10:25 Break

10:25

1aNS5. Modeling outdoor sound propagation in urban environments. Matthew Kamrath, Julien Maillard, Philippe Jean, Dirk Van Maercke (Ctr. Scientifique et Technique du Bâtiment (CSTB), CSTB, 24 Rue Joseph Fourier, Saint-Martin-d'Hères 38400, France, matthew.kamrath@cstb.fr), and Judicaël Picaut (LUNAM Université, Ifsttar, AME, LAE, Bouguenais Cedex, France)

Modeling outdoor sound propagation in cities is challenging because they often have complicated geometries and large domain sizes. One hybrid approach uses an engineering method (e.g., Harmonoise or Nord2000) with an extra attenuation term for interactions with complicated geometries like a T-barrier. The extra term quantifies the additional attenuation from the complex geometry compared to a reference geometry that can be modeled in the engineering method. Calculating pressure levels using a detailed method such as the Boundary Element Method (BEM) for the complex and reference geometries over multiple source and receiver positions and frequencies yields a table of excess attenuations. Interpolating this table of corrections at the desired source/receiver location and frequency produces the desired relative attenuation. This presentation discusses the development of this method for a case that has multiple propagation paths due to reflections from buildings and one interaction with a complex geometry per path but no additional diffraction points. The hybrid method agrees well with 3D BEM calculations at low frequencies.

10:45

1aNS6. Toward quantifying community response to crackle-containing jet waveforms. Kent L. Gee, Aaron B. Vaughn, Tracianne B. Neilsen, Kyle G. Miller (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), Michael M. James, Alexandria R. Salton, and J. M. Downing (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Although the phenomenon referred to as "crackle" has been previously described to be a dominant and annoying component of high-power military jet noise, its actual subjective impact is poorly understood. One of the challenges in quantifying jet crackle has been the identification of suitable metrics that are sensitive to the randomly occurring acoustic shocks responsible for the crackle percept [Gee *et al.*, J. Acoust. Soc. Am. **121**, EL1–EL7 (2007)]. This paper describes why understanding crackle could influence community perception of jet noise and recent waveform analyses of military jet noise that may provide insights how the phenomenon can be quantified perceptually. [Work supported by the USAFRL SBIR program.]

11:05

1aNS7. The reduction of gunshot noise and auditory risk through the use of firearm suppressors. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), Michael Stewart (Commun. Disord., Central Michigan Univ., Mount Pleasant, MI), Gregory A. Flamme, Stephen M. Tasko (Speech Pathol. and Audiol., Western Michigan Univ., Kalamazoo, MI), James E. Lankford (Allied Health and Commun. Disord., Northern Illinois Univ., Dekalb, IL), and Deanna K. Meinke (Audiol. and Speech Sci., Univ. of Northern Colorado, Greeley, CO)

Law enforcement, security, and military personnel train with small-caliber firearms that present a significant risk of noise induced hearing loss for the operator and range instructors. Measurements of three rifles and one pistol equipped with suppressors were conducted at an outdoor firing range using subsonic and supersonic ammunition. Suppressed and unsuppressed recordings were analyzed. Microphones were located to the left of the muzzle, to the right and left of the shooter's head, and one meter behind the shooter's head at the nominal instructor's position. Recordings were collected with a National Instruments PXI 1082 chassis with an NI 4499 data acquisition board at a 200 kHz sampling rate. Analysis of the peak sound pressure levels (dB SPL) and 8-h equivalent A-weighted energy (LAeq8) were conducted. The suppressors reduced the peak between 15 and 25 dB SPL and the LAeq8 between 8 and 28 dB. Reduced noise levels at the source will reduce auditory risk but do not necessarily eliminate the need for hearing protection. An outdoor noise propagation model based on ISO 17201-1 will be used to investigate the effect of suppressors to reduce noise in surrounding communities.

11:25

1aNS8. The effects of horizontal shading device orientation and surface material and the use of a compact silencer in forming acoustical barriers for a ventilated glass double skin facade. Jeehwan Lee, Jae D. Chang, and Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, coffeen@ku.edu)

Acoustical discomfort due to traffic noise transmission via operable windows is a significant environmental stress for building occupants in urban areas. A ventilated double skin facade, DSF, can potentially address the issue of noise transmission and operable windows. A hypothetically designed glass DSF mock-up was constructed in the opening between the reverberation rooms at the University of Kansas to determine the noise reduction of the DSF mock-up based on the orientation and surface materials of the horizontal shading devices and the use of a compact silencer in place of a typical ventilation grille in the outer skin opening. The compact silencer provides natural ventilation by allowing outside air to flow within the cavity of the DSF. For these laboratory tests, the horizontal shading devices with sound absorbing material were oriented at 0, 30, 60, and 90 degree angles. The laboratory noise reduction measurements are presented for the mock-up with the several shading device orientations and materials and for the use of a compact silencer.

MONDAY MORNING, 23 MAY 2016 SALON H, 8:15 A.M. TO 12:00 NOON

Session 1aPA

Physical Acoustics, Structural Acoustics, and Vibration, and Engineering Acoustics: Computational Methods in Physical Acoustics I

Amanda Hanford, Cochair
Applied Research Lab, Pennsylvania State Univ., PO Box 30 - MS 3230D, State College, PA 16804

D. Keith Wilson, Cochair
CRREL, U.S. Army ERDC, 72 Lyme Rd., Hanover, NH 03755-1290

Chair's Introduction—8:15

Invited Papers

8:20

1aPA1. Assessment of numerical accuracy for the direct computation of sound propagation. Roberto Sabatini (CEA, DAM, DIF, Ecully, France), Olivier Marsden (European Ctr. for Medium-Range Weather Forecasts, Ecully, France), Christophe Bailly, and Christophe Bogey (Ctr. Acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Ecole Centrale de Lyon, bat. KCA, 36 Ave. Guy de Collongue, Ecully 69134, France, christophe.bailly@ec-lyon.fr)

The computation of long-range sound propagation, which consists in solving the full 3-D compressible Navier-Stokes equations in order to include all physical phenomena correctly, is now feasible. These progress are mostly due to the development of high fidelity algorithms in aeroacoustics over the last two decades, characterized by a high spectral accuracy and a low selective dissipation. The

accuracy of these algorithms has been validated by considering various benchmark problems in computational aeroacoustics. However, assessment of the numerical accuracy remains tricky for nonlinear problems. In the context of long-range atmospheric propagation of infrasounds, nonlinear effects induced by large relative amplitudes and molecular absorption in the upper atmosphere can be significant. A direct computation of the propagation and a spectral analysis of the acoustic signals are here performed. The importance of the numerical dissipation is examined by considering the spectral transfer function of the numerical algorithm, including the shock-capturing scheme, which can be explicitly expressed for the optimized finite-difference schemes used in this work.

8:40

1aPA2. Eulerian methods for high frequency acoustics. Sheri Martinelli (Marine & Physical Acoust., ARL Penn State, P.O. Box 30, Mailstop 3230D, State College, PA 16804-0030, slm77@psu.edu)

The high frequency approximation to the wave equation yields an eikonal equation for the wave phase and a transport equation for the amplitude. Unfortunately, straightforward computation of the eikonal equation poses a challenge in that solutions may be multivalued which affects the convergence of a numerical scheme. Traditional numerical schemes invoke ray tracing methods to solve the eikonal equation in a Lagrangian frame of reference. Ray tracing performs well in many cases and handles the multivalued solution inherently. However, in long range simulations or simulations with complicated boundary conditions, ray solutions may perform poorly as rays are very sensitive to small perturbations in initial conditions. In the past few decades, several methods for solving the eikonal equation in a fixed (Eulerian) frame of reference have been devised. As with any computational method, each has its own advantages and disadvantages, but it is useful to be aware of alternatives since many problems in physical acoustics may benefit from having phase solutions computed on a fixed grid. We present a brief review of recent Eulerian approaches and discuss the merits and open issues with each. [Work supported by The Science, Mathematics, and Research for Transformation (SMART) Scholarship for Service Program.]

9:00

1aPA3. Eigenfunction analysis of the Green's function parabolic equation. Kenneth E. Gilbert (National Ctr. for Physical Acoust., Univ. of MS, P.O. box 35, 1703 Hunter Rd., Thaxton, MS 38871, kgilbert@olemiss.edu)

The Green's function parabolic equation (GFPE) algorithm was originally derived using operators, functional analysis, and Green's functions [K. E. Gilbert and X. Di, J. Acoust. Soc. Am. **94**, 2343–2352 (1993)]. This presentation demonstrates that the GFPE algorithm can be obtained more simply and directly in terms of the eigenfunctions of a homogeneous atmosphere over a complex impedance ground surface. The eigenvalue-eigenfunction equations obtained follow naturally from the basic and well-known separation of variables method. The eigenfunctions, which are fundamental solutions of the Helmholtz equation, consist of plane waves plus a surface wave. The plane waves have a continuous wave number spectrum while the surface wave has a single, discrete, complex wave number. A physical explanation is given for the existence of the surface wave. It is shown that eigenfunction expansions give concrete meaning to abstract operator solutions for one-way propagation. Using the notion of a function of an operator and an eigenfunction expansion of the acoustic field, the GFPE can be obtained simply and concisely.

9:20

1aPA4. Inverse methods for Green's function retrieval. Sandra L. Collier, Jericho E. Cain, John M. Noble, David A. Ligon, W. C. Kirkpatrick Alberts, and Leng Sim (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil)

The Green's function, or medium impulse response, provides information on the propagation channel. Here, we develop different inverse methods, based on time-reversal and flow-reversal methods, to retrieve the Green's function using audible acoustic sources. These methods are then applied to experimental data collected using four 12-element, tri-axis, microphone arrays. Initial findings indicate that Green's function retrieval is feasible.

9:40

1aPA5. A spacetime finite element method for coupled acoustic fluid-structure interactions. Scott T. Miller (Computational Solid Mech. & Structural Dynam., Sandia National Labs., P.O. Box 5800, MS 0845, Albuquerque, NM 87185, stmille@sandia.gov) and Amanda D. Hanford (Marine and Physical Acoust., Pennsylvania State Univ., State College, PA)

The spacetime discontinuous Galerkin (SDG) finite element method is applied to problems of acoustic propagation. The spacetime discontinuous Galerkin are a family of discontinuous finite element methods for hyperbolic systems of equations. They utilize an advancing front mesh generation procedure that allows local "patches" of elements to become decoupled from the global solution domain through causality. Causal space-time meshes enable a locally implicit solution procedure with provably linear computational complexity. Mesh adaptivity is shown to effectively resolve propagating waves and minimize dispersion error. Numerical results demonstrate the effectiveness of the proposed solution method, including achieving optimal convergence rates. Time domain simulations of several canonical acoustic problems are presented.

10:00–10:20 Break

10:20

1aPA6. Uncertainty quantification, turbulent scattering, and random sampling in outdoor sound propagation calculations. D. K. Wilson (U.S. Army Cold Regions Res. Eng. Lab., U.S. Army Engineer Res. Dev. Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D. Keith.Wilson@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), Carl R. Hart, and Vladimir E. Ostashev (Cold Regions Res. Eng. Lab., U.S. Army Engineer Res. Dev. Ctr., Hanover, NH)

Although the fundamental physics of sound propagation outdoors (e.g., absorption, ground interactions, refraction, and scattering) are well understood, modeling capabilities are typically inadequate to capture the many intricate interactions with the atmosphere, ground, and natural and man-made terrain features. Furthermore, environmental data are rarely available at the sub-wavelength resolution needed to accurately predict the sound field in the audible range, in a deterministic sense. Fortunately, advances in computing capabilities and statistical sampling methods provide new opportunities for quantifying the inherent uncertainties of sound propagation predictions. Among the sampling techniques considered in this presentation are ordinary Monte Carlo sampling, Latin hypercube sampling, stratified sampling based on meteorological classes, importance sampling, and adaptive importance sampling. When uncertainties in the atmospheric and ground properties dominate, importance sampling is found to converge to an accurate estimate with the lowest calculation time. When random turbulent scattering dominates, the sampling method has little impact. Efficient and accurate approaches to calculating sound level statistics and quantifying uncertainty result from simultaneously sampling over frequency, source/receiver position errors, uncertain atmospheric profiles and ground properties, and random processes such as scattering by turbulence and vegetation.

10:40

1aPA7. Development of a simulation for supersonic aircraft signature propagation through turbulence. Trevor A. Stout and Victor W. Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, tqs5346@psu.edu)

For the next generation of quiet supersonic civilian aircraft, minimizing the acoustic signature is important. To inform the design process, a model of the effect of shock wave propagation through the atmosphere is required. For such a model to be accurate, both the weak nonlinear effects on the supersonic signature and the distortion due to atmospheric turbulence could be important. To best predict the variability of waveforms heard at multiple locations on the ground, a propagation simulation requires multiple random realizations of the turbulent field. In the present model, a finite difference implementation of the two-dimensional, nonlinear KZK propagation equation is combined with a statistical turbulence model to simulate shock wave propagation through the atmosphere. For this initial work, a KZK simulation for focused sources developed at the University of Texas by Y. Lee was modified and extended for atmospheric propagation. [Work supported by NASA via a subcontract from Wyle.]

Contributed Papers

11:00

1aPA8. Sound synthesis methods of fire flame sound for predicting hidden images. HyunIn Jo (Architectural Eng., Hanyang Univ., Seoul, Seongdong-gu 133-791, South Korea, best2012@naver.com), Rehan H. Afzal, Imran Muhammad (Architectural Eng., Hanyang Univ., Seoul, Pakistan), and Jin Y. Jeon (Architectural Eng., Hanyang Univ., Seoul, South Korea)

A methodology is proposed and investigated for realistically synthesizing aerodynamic sounds, such as fire sounds based on a graphically generated animation of fire. The proposed technique, "bandwidth extension method," is a physically based combustion sound model for synthesizing low frequency flame sounds from a visual flame simulation, which runs at low temporal sampling rates. The spectral bandwidth approach uses noise matching combustion sound spectra, whereas the data-driven texture approach uses input flame sound recordings. Various simulations of fire animations are presented, and a comparison is provided of small- and large-scale phenomena for plausible synthesized and recorded flame sounds of various flames.

11:15

1aPA9. Modeling and experimental validation of an evanescent wavefield using a wavenumber integration method. Daniel Plotnick (APL, Univ. of Washington, 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com) and Philip L. Marston (Washington State Univ., Pullman, WA)

When sound is incident on the boundary between two media, where the first medium has a lower sound speed than the second (such as sound in water incident on some sediments), a critical angle exists. If the incident angle is below this critical angle an evanescent wavefield is formed in the second medium. Experimental work measured the structure of such a wavefield at an oil-water interface insonified by an appropriately oriented transducer, where the region of interest is not in the transducer far-field [Osterhoudt *et al*., IEEE J. Ocean. Eng. **33**, 397–404 (2008)]. Several features of note within the wavefield were observed. This work attempts to model those observations using wavenumber integration techniques. In the incident plane wave approximation, the evanescent wavefield is vertically phase-locked and the characteristic exponential spatial decay is constant across the wavefield. However, the contribution of many wavenumber components across the region of interest is key to understanding the wavefield structure seen in experiments. This model will be compared to the experimental results, and the details of phase, amplitude, and the overall wavefield structure discussed. An improved approximation of the transducer orientation used in experiments may also be inferred from these results. [Work supported by ONR.]

11:30

1aPA10. A modal time-domain method for modeling dispersive and dissipative linear waves. Jonathan Botts (Aalto Univ., 209 N. Commerce St., Ste 300, Culpeper, VA 22701, botts.jonathan@gmail.com) and Lauri Savioja (Aalto Univ., Espoo, Finland)

A modal time-domain method is developed for simulating linear dispersive and dissipative waves in 3D geometries. Two of the principle challenges in modeling large acoustic systems are fitting models into computer memory and managing numerical wave speed errors; both prevent modeling of large problems and high frequencies. To address these issues, a framework is developed for solving linear wave equations on a structured subdivided domain using exact frequency-dependent time integration. From the perspective of modeling sound in rooms, the method is shown to accurately capture propagation losses due to air absorption, dispersive bending waves in panels, lossy propagation in porous media, and dispersionless propagation over large distances. This approach enjoys exact time integration for many PDE, but such accuracy comes at the cost of highly-structured grids and non-trivial boundary conditions. Advantages and shortcomings are discussed relative to problems of practical size and interest.

11:45

1aPA11. Design and evaluation of a new ultrasonic horn for viscosity reduction of heavy oil. Yan Yang, Xiuming Wang, and Delong Xu (Inst. of Acoust.,Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100000, China, xudelong@mail.ioa.ac.cn)

Based on the basic principle of the sandwich piezoelectric transducer, a high-power ultrasonic horn which can be used for viscosity reduction of heavy oil and pour point reduction of crude oil with high pour point was proposed and designed. The basic frequency of the horn is 16.8 kHz, and the input electrical power is 1 kw. It can work continuously on the industrial scale. First, the impedance characteristic was derived for the simplified model of the sandwich piezoelectric horn on the basis of equivalent network method. Second, its properties such as the resonance frequency, conductance, and transmitting voltage response were simulated by using of ANSYS-software in air, water, and silicone oil, respectively. According to the simulation, a prototype was made and measured by an impedance analyzer. Its basic frequency is in good agreement with the simulation calculation. The result shows that this high-power ultrasonic horn is expected to be applied on the industrial scale in viscosity reduction of heavy oil and ultrasonic sludge disintegration.

MONDAY MORNING, 23 MAY 2016 — SALON E/F, 8:00 A.M. TO 12:00 NOON

Session 1aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Potpourri (Poster Session)

Alan Kan, Chair

University of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705

Posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to view 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.

Contributed Papers

1aPP1. Investigating the role of the medial olivocochlear reflex in selective attention. Jordan A. Beim, Andrew J. Oxenham, and Magdalena Wojtczak (Psych., Univ. of Minnesota, N218 Elliott Hall, 75 East River Rd., Minneapolis, MN 55455, beimx004@umn.edu)

Selective attention can dramatically attenuate the cortical responses to unattended stimuli relative to attended stimuli. The mechanisms behind this attenuation and their locus in the auditory pathways are not yet understood. Recent animal work suggests that changes may begin to occur as early as the cochlea itself, potentially through the medial olivocochlear (MOC) reflex. Since measuring MOC activity directly is too invasive to be done in humans, otoacoustic emissions (OAEs) are used to examine changes in cochlear function. Previous investigations of attention-related efferent effects on cochlear processing using OAEs have generally found small and inconsistent effects. The previous lack of effects could be due in part to shortcomings in OAE and MOC-reflex measurement methodology. A new experimental paradigm designed to overcome these shortcomings was used here to examine the effects of selectively attending to visual or auditory stimuli. Participants directed attention to either low-frequency tones, high-frequency tones, or visual stimuli to complete a task with each stimulus while OAEs were recorded using the same low-frequency tones. Initial results have revealed an average decrease in emission magnitude of 8.5 dB when participants attended to the visual stimulus compared with the low-frequency tones, suggesting substantial attentional effects on cochlear responses.

1aPP2. Investigation of the relationship between cochlear gain reduction and speech-in-noise performance at positive and negative signal-to-noise ratios. Kristina D. Milvae, Joshua M. Alexander, and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, kderoy@purdue.edu)

The ability to understand speech in noisy environments is vital to effective communication, but the auditory mechanisms driving this ability are not well understood. The auditory system may decrease the response to stimulated frequency regions so that changes from the acoustic environment are emphasized and information transfer is maximized. A known physiological mechanism that may contribute to this ability is the medial olivocochlear reflex (MOCR). The MOCR reduces cochlear gain via a neural feedback pathway between the brainstem and cochlea. A forward masking paradigm was used to examine cochlear gain reduction of a 2-kHz tone, an important frequency for speech perception. Speech-in-noise performance was also measured for the same participants at positive and negative signal-to-noise ratios (SNRs). Correlational analysis was used to examine the relationship between cochlear gain reduction and speech-in-noise performance across SNR. It was hypothesized that the correlation would depend on SNR, because turning down the gain should only improve performance when the overall response to the noise is decreased more than the response to the speech. Results and implications will be discussed. [Research supported by NIH(NIDCD) T32 DC000030.]

1aPP3. Evaluation of a two-interval, observer-based behavioral procedure for assessing detection performance in 2- to 4-year-old children. Lori Leibold (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68124, lori.leibold@boystown.org), Angela Y. Bonino, Nicole Corbin, and Emily Buss (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Collecting reliable behavioral data from toddlers and preschoolers is challenging, limiting our understanding of human auditory development for these age groups. This study evaluated an observer-based procedure for measuring detection thresholds using a two-interval, two-alternative forced-choice paradigm. Listeners were normal-hearing adults and 2- to 4-year-olds. Detection performance was assessed in one of three conditions: (1) 1000-Hz warble tone in quiet; (2) disyllabic word in speech-shaped noise; or (3) disyllabic word in two-talker speech. Listeners were trained to perform a play-based, motor response whenever they heard a signal (e.g.,

putting a block in a bucket). An experimenter observed the child's behavior and judged whether the signal was presented during the first or second observation interval; the experimenter was blinded to the true signal interval, so this judgment was based solely on the child's behavior. Following training, signal level was adjusted to estimate 71% correct detection. The masker, when present, was played continuously throughout testing at a level of 55 dB SPL. The yield rate was 100% for adults and >80% for children. Good reliability was seen within a track and across testing sessions for individuals, indicating the two-interval procedure is feasible and reliable for use with toddlers and preschoolers.

1aPP4. Effects of various acoustic stimuli on higher order thinking and tinnitus perceptions. Andrew Cobabe (Audiol., Intermountain Healthcare, 1350 North 500 East, Logan, UT 84341, acobabe@gmail.com), Karen Munoz, Don Sinex, and Jeffery Larsen (Communicative Disord. and Deaf Education: Audiol., Utah State Univ., Logan, UT)

Masking sounds are often employed in the management of subjective tinnitus to desensitize and divert attention. The potential side effects of these maskers in distraction from other thought processes is not well understood. This research addresses the effects an artificial tinnitus and broadband masking or distracting stimuli have on performance of challenging cognitive or attention tasks. Masking sound preferences were also assessed. Adults with normal hearing and without thought processing disorders performed a two-back (n-back) auditory memory task in listening conditions including silence, white noise, classical music, pop music, and an artificial tinnitus condition. Participants also completed a survey, rating perceptions of their performance and attentional preferences for the sounds. Data show a greater distraction effect for more complex stimuli (pop music). Natural environmental sounds or white noise are favored to pop music for tinnitus masking. It is suggested that tinnitus maskers with more complex meaning including singing, language, and speech appear to have a more multifaceted effect on attention. Although this may be desirable for effective tinnitus masking, the impact on other thought processes appear to be present but unintended. The irrelevant sound effect (ISE) of tinnitus maskers warrants further investigation.

1aPP5. A model of the cochlear apex and structure for study of mechanism and response characteristics. Samiya Alkhairy (MIT, Eaton Peabody Lab, MEEI 243 Charles St., Boston, MA 02114, samiya@mit.edu) and Christopher Shera (EPL-MEEI, Boston, MA)

We provide a model of the cochlear apex that is geared towards studying: cochlear function represented by common response characteristics (e.g., sharpness of tuning), governing mechanisms, inter-relations between the response characteristics, inter-relations between the traveling wave and point admittance effective mechanics, and the relationship between response characteristics and mechanism. The model currently used for some of these ends is an uncoupled resonant simple harmonic oscillator. However, this framework is inconsistent with current knowledge in the field. Hence, an alternative framework must be provided for these purposes. To these ends, in this paper, we construct a general parametric model of the cochlear that represents the Organ of Corti as a single partition in a box fluid model. The model provides closed-form expressions for responses—specifically membrane velocity and fluid pressure—and the corresponding response characteristics. Development of the model includes construction of effective mechanics, construction of responses and their characteristics, and validation. We find that our model provides a suitable representation of data, by using auditory nerve fiber Wiener-Kernel data to approximate mechanical responses. Also, we show that the model may be extrapolated beyond its primary purposes for use as a computationally efficient cochlear front-end to auditory aid devices and speech processing programs.

1aPP6. Examination of middle-ear function through interspecies comparison of middle-ear transmission characteristics. Charlsie Lemons and Julien Meaud (Georgia Inst. of Technol., 1035 Hampton St., Atlanta, GA 30332, charlsielem@gmail.com)

Lumped parameter ossicular chain models for several mammals have been developed; by comparing model parameters and the input and output characteristics of these models, conclusions can be drawn about the relative middle-ear transmission characteristics between different species allowing for a better understanding of the middle-ear's overall role in hearing. The tympanic membrane plays a significant role in middle-ear function: experiments show that the forward and reverse transmission characteristics of the middle-ear are affected by the tympanic membrane, especially at high frequencies. Thus, understanding the behavior and transmission characteristics of the tympanic membrane is vital in understanding overall transmission through the middle-ear. By examining the interaction between lumped parameter ossicular chain models and tympanic membrane models from several animals and by comparing normalized two-port matrix parameters from several species, observations regarding differences in middle-ear behavior between species are drawn.

1aPP7. Electrically-evoked frequency following responses in the auditory brainstem of cochlear impaired guinea pigs. Jing Chen (Speech and Hearing Res. Ctr., Key Lab. of Machine Percept. (Ministry of Education), Ctr. for Information Sci., Peking Univ., Rm. 2228, No. 2 Sci. Bldg., Beijing 100871, China, chenj@cis.pku.edu.cn), Xiuyong Ding (Dept. of Otorhinolaryngology Head and Neck Surgery, Beijing Xuanwu Hospital, Capital Medical Univ., Beijing, China), Wenxin He (Speech and Hearing Reseach Ctr., Key Lab. of Machine Percept. (Ministry of Education), Peking Univ., Beijing, China), Ruxiang Zhang (Dept. of Otorhinolaryngology Head and Neck Surgery, Beijing Xuanwu Hospital, Capital Medical Univ., Beijing, China), and Xihong Wu (Speech and Hearing Reseach Ctr., Key Lab. of Machine Percept. (Ministry of Education), Peking Univ., Beijing, China)

It is still a difficult clinical issue to decide whether a patient is a suitable candidate for a cochlear implant and to plan postoperative rehabilitation, especially for some special cases, such as auditory neuropathy. A partial solution to these problems is to preoperatively evaluate the functional integrity of the auditory neural pathways. For evaluating the strength of phase-locking of auditory neurons, which was not reflected in previous methods using electrically evoked auditory brainstem response (EABRs), electrically evoked frequency following responses (EFFRs) recorded on normal guinea pigs have been studied in a previous study (He *et al.* 2014, PLoS One). To investigate the feasibility of EFFRs for clinic, a cochlear impaired model of guinea pig was used to compare the EFFRs for normal and impaired hearing, and to compare EFFRs and EABRs for the impaired hearing. Eight guinea pigs were used, and the experiment results showed that: (1) there were no significant differences of EFFRs between the normal and the cochlear impaired guinea pigs on the relative amplitudes, the latencies, the frequency ranges, and the evoked thresholds; (2) the evoked threshold of EFFRs was lower than EABRs, and the EFFRs were recorded successfully more times than EABRs.

1aPP8. Effect of asymmetrical organ of Corti mechanics on cochlear fluid pressure patterns. Wenxiao Zhou and Jong-Hoon Nam (Mech. Eng., Univ. of Rochester, 212 Hopeman Bldg., Rochester, NY 14627-0132, jong-hoon.nam@rochester.edu)

In the cochlea, the sensory epithelium called the organ of Corti (OC) vibrates due to hydrodynamic pressures. The cochlear fluid pressure can be divided into differential and average components. The differential fluid pressures across the elastic OC complex (the OC, the tectorial membrane, and the basilar membrane) generate slow traveling waves toward the apex of the cochlea. Theoretical studies have assumed that the differential pressure component is relevant to cochlear mechano-transduction because the average component cannot vibrate the OC complex. This assumption has an implication that the entire section of the OC complex vibrates in synchrony. That is, the basilar membrane can represent the sole fluid-interacting surface of the OC complex. However, several recent studies considered that the active motility of the outer hair cell can break the synchrony between the top and the bottom fluid-interacting surfaces of the OC complex. Using a computational cochlear model with fully deformable OC complex, we present how different vibration modes of the OC complex can affect the cochlear fluid pressure patterns. The intra-cochlear fluid pressure patterns of our model are compared with recent measurement results. A condition under which the average pressure component can affect cochlear mechano-transduction is discussed.

1aPP9. Effect of sub-tectorial space fluid dynamics on cochlear energy dissipation and inner hair cell mechano-transduction. Srdjan Prodanovic, Sheryl M. Gracewski, and Jong-Hoon Nam (Mech. Eng., Univ. of Rochester, 212 Hopeman Bldg., Rochester, NY 14627-0132, s.prodanovic@rochester.edu)

In the mammalian cochlea, the mechano-transduction of the inner hair cell (IHC) stereocilia occurs in a micrometer-thick fluid space between the tectorial membrane and the reticular lamina. Using a computational model of the cochlea, we analyzed how the sub-tectorial space (STS) fluid mechanics affect cochlear power dissipation and IHC mechano-transduction. Based on the simulations of a single IHC stereociliary bundle in the STS (Prodanovic *et al*., 2015), the fluid-induced forces of the STS were reduced to simple equations. The reduced STS response was combined with a whole cochlear model consisting of: organ of Corti structural mechanics, cochlear fluid dynamics, and outer hair cell electro-physiology. Energy dissipation in the cochlea was quantified for three categories: macro-fluidic dissipation along the cochlear scalae, micro-fluidic dissipation in the STS, and other dissipation in the organ of Corti. The phase of IHC mechano-transduction current with respect to the basilar membrane displacement was dependent on stimulating frequency and location.

1aPP10. Functional characterization of auditory forebrain in the songbird. Taffeta M. Elliott (Psych., New Mexico Inst. of Mining and Technol., 801 Leroy Pl., Socorro, NM 87801, elliott@nmt.edu) and Frédéric E. Theunissen (Psych., Univ. of California, Berkeley, Berkeley, CA)

Songbirds communicate with individualized vocalizations that must be discriminated and recognized based on the detection of complex sound features. Neurons in the primary auditory nucleus of the zebra finch, field L, specialize in extracting auditory features that include rapid fluctuations in temporal envelope, harmonic structure, and dynamic pitch changes. We used electrophysiological recording and immunohistochemistry to determine the inhibitory network roles and putative connectivity of field L neurons that represent specific kinds of acoustic features. Neuronal responses to conspecific song broadcasts were recorded juxtacellularly from an *in vivo* anesthetized preparation. At the end of recordings sufficient to calculate a spectrotemporal receptive field (STRF), the cell was filled with Neurobiotin from the patch electrode. Some projections could be traced. A slight majority of recovered auditory neurons were GABA-ergic based on double fluorescent labeling of the Neurobiotin and an antibody for glutamic acid decarboxylase (GAD). The STRFs of the recovered neurons showed a diverse sampling of complex and simpler types of auditory selectivity. This variety should be useful in distinguishing signature features of particular individuals' songs. We hypothesize that uneven distribution of tuning across morphological types and anatomical locations superficial within field L contributes to parallel pathways dedicated to processing distinct categories of acoustic features.

1aPP11. A discrete time parametric model of auditory filters at peak sensitivity. Christopher L. Sullivan (Elec. Eng., Univ. of Illinois, 203 S Coler St., Apt. 2, Urbana, IL 60801, sulliv45@illinois.edu) and Jont Allen (Elec. Eng., Univ. of Illinois, Mahomet, IL)

The first stage of human auditory processing is the filtration performed by the ear. The middle ear, basilar membrane, and tectorial membrane contribute to a filter bank specialized for amplitude and frequency selectivity. Accurately modeling this system is of critical importance, because the properties of the cochlear filter bank determine what audio information is available to humans. Because the cochlea is such an intricate and non-linear system, many cochlear models are computationally taxing and unusable for real time applications. By making a judicious linear approximation of the system, digital filters modeled after the cochlea can be implemented efficiently in real time. However, the digital filter models presently in use (often eighth order gammatone or elliptic filters) do a poor job of capturing the most perceptually critical properties of the cochlear filter bank. In this report, a set of parametric digital filters are proposed, which accurately model cochlear filters at peak sensitivity from 80 Hz to 19.5 kHz best frequency. The filters were designed using a gradient of steepest descent method to fit target frequency responses generated by a physical model of the cochlea. The final result is a set of polynomial equations which describe the locations of the poles and zeros of the digital cochlear filter approximations as a function of normalized cochlear best place.

1aPP12. Cortical oscillatory signatures of active listening complement brainstem coding measures in predicting listening performance. Hari M. Bharadwaj (Athinoula A. Martinos Ctr. for Biomedical Imaging, Massachusetts General Hospital, 149 Thirteenth St., Boston, MA 02129, hari@nmr.mgh.harvard.edu), Inyong Choi (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), and Barbara Shinn-Cunningham (Biomedical Eng., Boston Univ., Boston, MA)

Following the finding of cochlear deafferentation ("synaptopathy") in noise exposure and aging in animal models (Kujawa and Liberman, J. Neurosci. 2009; Sergeyenko *et al*., J. Neurosci. 2013), there is considerable interest among hearing scientists and clinicians to investigate the presence and consequences of synaptopathy in humans. In the laboratory, human subjects with clinically normal hearing (NH) listeners from the general population exhibit large individual differences in suprathreshold perceptual ability. Through a recent series of experiments using otoacoustic emissions and electrophysiology, we studied these individual differences and showed that they partly arise from differences in subcortical coding of temporal information consistent with synaptopathy (Bharadwaj *et al*., J. Neurosci. 2015). In the current study, we examined cortical oscillations accompanying cued preparation to selectively attend to a target spatial location as a correlate of top-down active listening. We find that both brainstem envelope-following responses (EFRs) and cortical oscillations independently correlate with performance. EFRs and cortical oscillations together explain a greater fraction of the variance across individuals than EFRs alone. Our results point to a mechanism by which some listeners with good subcortical coding may still perform poorly, and a technique that might be employed to isolate sensory coding contributions to suprathreshold hearing deficits.

1aPP13. Frequency cueing and top-down processes in the perceptual segregation of simultaneous tone. Yi Shen and Brooke E. Louthan (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S Jordan Ave., Bloomington, IN 47405, shen2@indiana.edu)

Discriminating the properties of two simultaneously presented pure tones could be challenging even when the two tones are distant in frequency, potentially due to the tones being perceptually grouped into a single auditory object. In the current study, two tones 1.1 octaves apart were presented monaurally into the same ear, only one of which was amplitude-modulated at a depth of 100% and a rate of 10 Hz. Listeners selected the amplitude-modulated tone from the pair, and performance thresholds were estimated in terms of the presentation duration required to carry out the task. Moreover, the experiment included conditions in which the frequency of one of the tones was cued by a precursor tone, presented to the ipsilateral or contralateral ear. For ipsilateral precursors, when the precursor was relatively long (400 ms), the presence of precursor improved threshold from the no-precursor condition independent of the gap duration between the precursor and the tone-pair. On the other hand, when the precursor was relatively short (100 and 200 ms), the threshold improvement was relatively small for short gap duration. The facilitation effect of the precursor was observed for gap duration as long as 1 s, suggesting the involvement of top-down processes. For contralateral precursors, the facilitation effect was much reduced.

1aPP14. Age-related effects on two-tone suppression and consonant perception in noise. Erica L. Hegland, Alexander L. Francis, and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci. Dept., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu)

Two-tone suppression reduces gain in the cochlea nearly instantaneously in a frequency-dependent manner. In speech perception, suppression may enhance spectral contrasts between regions of higher and lower energy. A few previous studies have investigated the effects of aging on suppression and correlations with speech perception in noise, but none have looked at the adaptability of suppression with preceding stimulation. In the present study, estimates of two-tone suppression and consonant perception were measured in younger and older adults. Suppression was measured with short tonal stimuli with and without preceding stimulation. Consonant vowel (CV) stimuli consisted of combinations of the consonants /b, g, d/ and the vowels /ɑ, i, u/ spoken by three male speakers. Broadband noise was filtered to match the long-term average spectrum of the speech stimuli used. CV onset was either 0 or 70 ms after onset of the noise. Participants were asked to identify the consonant by choosing /b/, /g/ or /d/. Percentage correct was calculated for CVs in quiet and in +5, 0, −5, and −10 dB signal-to-noise ratio conditions. Suppression and adaptability were found to correlate with several aspects of speech perception in noise. Age-related results and implications will be discussed. [Research supported by NIH(NIDCD)F31 DC014395.]

1aPP15. Comparison of different procedures to assess spectral weights for speechlike sounds. Jennifer Lentz and Kimberly Skinner (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu)

We present a spectral weighting experiment designed to assess whether listeners adopt different listening weights for consonant sounds depending on the sound presented and whether weights depended on the level of analysis—for the individual stimuli ("microscopic") level or for the stimulus set ("macroscopic"). We used a closed set of speech stimuli modeled after the voiceless fricatives /f/, /s/, and /ʃ/. Subjects heard one of the three stimuli, each containing random level perturbations across multiple frequency bands. They then made an identification decision. For each stimulus, we estimated the weighting strategy by correlating observers' responses with the random perturbations. This work allows us to estimate the strategies applied to each stimulus in the set. This method differs from other speech weighting studies, which assume that listeners apply a general filter to speech stimuli, enhancing some bands and attenuating others (e.g., the Articulation Index). We will also compare our data to those using a global strategy to determine the importance of analyzing decisions on an individual-stimulus basis. Our data indicate that the listening strategy depends greatly on the stimulus presented to them, and that we can assess the flexibility of the auditory system in adapting to changing stimuli using these methods.

1aPP16. Infants' ability to separate concurrent vowels based on voice characteristics. Lynne Werner and Monika Oster (DSpeech & Hearing Sci., Univ. of Washington, 1417 North East 42nd St., Seattle, WA 98105-6246, lawerner@u.washington.edu)

The ability to use voice characteristics to identify a target vowel in the presence of competing vowels was assessed in normal hearing 3- and 7-month-old infants and adults. Listeners heard two trains of simultaneously presented vowels, one spoken by a male, the other spoken by a female. The nine vowels were 400 ms in duration, with 1200 ms between vowels, presented at a level of 70 dB SPL. Listeners learned to respond when a target vowel was spoken by a target talker, but to neither other vowels or the target vowel spoken by the non-target talker. Each age group was able to respond selectively on target trials, with d' > 1; the response rates to no-target and probe trials were not different at any age. However, only about 40% of infants achieved at least 80% responses on target trials with no more than 20% responses on no-target trials, and only about half of those infants also achieved no more than 20% responses on probe trials within up to 40 trials. Thus, infants as young appear to be able separate speech sounds on the basis of voice characteristics. However, few infants achieve a high level of performance in this task.

1aPP17. Complex pitch perception via vocoder simulations of cochlear-implant processing. Anahita H. Mehta and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, N625, Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, mehta@umn.edu)

Melodic pitch perception in cochlear implants (CIs) is limited at least in part by spectral resolution, which in turn is limited by the number of spectral channels as well as interactions between adjacent channels. Future technical improvements may lead to greater numbers of functional channels within CIs; however, the target number of channels required to extract the fundamental frequency from a harmonic sound is not known. The present study used noise-vocoded simulations of CI processing to parametrically study melodic pitch perception, with other potential cues, such as temporal-envelope rate and lowest spectral component, removed. The number of spectral channels was parametrically varied, along with the degree of channel interaction. In contrast to earlier studies, preliminary results suggest that 32 channels are not sufficient to elicit the perception of complex pitch, even in the absence of any significant channel interactions. Even when pitch perception was possible with higher numbers of channels, performance was degraded by introducing varying amounts of channel interaction. The results suggest that spectrally based pitch is unlikely to be generated in CI users without a substantial change in technology and/or site of stimulation. [Work supported by NIH grant R01DC005216.]

1aPP18. A detection-theoretic analysis of the relation between multitone auditory streaming and masking. An-chieh Chang, Robert A. Lutfi, and Jungmee Lee (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, 1975 Willow Dr., Madison, WI 53706, anchieh.chang@wisc.edu)

Research on hearing has long been challenged with understanding our exceptional ability to "hear out" individual sounds in a mixture (the so-called cocktail party problem). Two general approaches to the problem have been taken using sequences of tones as stimuli. The first has focused on our tendency to hear sequences, sufficiently separated in frequency, split into separate cohesive streams (auditory streaming). The second has focused on our ability to detect a change in one sequence, ignoring all others (auditory masking). The two phenomena are clearly related, but that relation has never been specifically evaluated. The present paper offers a detection-theoretic analysis of the relation between multitone streaming and masking that underscores the expected similarities and differences between these phenomena and the predicted outcome of experiments in each case. The underlying premise of the analysis is that streaming is auditory system's way of maximizing the likelihood that sounds from separate sources are perceived as separate. Experiments are reported supporting some basic outcomes of the analysis.

1aPP19. Release from masking in cochlear-implant listeners and its relationship to spectro-temporal resolution. Naomi B. Croghan and Zachary M. Smith (Res. and Technol. Labs, Cochlear Ltd., 13059 East Peakview Ave., Centennial, CO 80111, ncroghan@cochlear.com)

Release from masking is commonly observed in normal-hearing listeners when fluctuating interferers mask speech perception less than steady interferers at the same signal-to-noise ratio. In contrast, studies of cochlear-implant listeners have historically failed to find masking release with fluctuating maskers. Here, we provide new insights into the relationship between spectro-temporal resolution and release from masking in cochlear-implant and simulated cochlear-implant listeners. Varying degrees of current spread were simulated in normal-hearing listeners using a tone vocoder. Speech reception thresholds were measured with four different interferers: a tone complex, speech-shaped noise, four-talker babble, and a single competing talker. A dynamic spectral ripple detection test was conducted to measure psychophysical spectro-temporal resolution. Results show that overall speech perception was correlated with spectro-temporal ripple detection thresholds. However, some maskers were more sensitive to spectro-temporal resolution than others. Subjects in both groups demonstrated release from masking between the speech-shaped noise and the single competing talker conditions, and the degree of release varied with spectro-temporal ripple sensitivity. These findings shed new light on previous cochlear-implant results, suggesting that individual cochlear-implant listeners with high spectro-temporal resolution may benefit from the temporal fluctuations in certain noise types.

1aPP20. Measuring vowel percepts in human listeners with behavioral response-triggered averaging. W. Owen Brimijoin (MRC Inst. of Hearing Res., Glasgow, United Kingdom), Emily Tilbury (School of Eng., Univ. of Glasgow, New Acoust., 1 Aurora Ave., Queens Quay, Clydebank G81 1BF, United Kingdom, emily@newacoustics.co.uk), Michael A. Akeroyd (MRC Inst. of Hearing Res., Glasgow, United Kingdom), and Bernd Porr (School of Eng., Univ. of Glasgow, Glasgow, United Kingdom)

A vowel can be largely defined by the frequencies of its first two formants, but the absolute frequencies for a given vowel vary from talker to talker and utterance to utterance. Given this variability, it is unclear what criteria listeners use to identify vowels. To estimate the vowel features for which people listen, we adapted a noise-based reverse-correlation method from auditory neurophysiological studies and vision research (Gold *et al.*, 1999). Listeners presented with the stimulus, which had a random spectrum with levels in 60 frequency bins changing every 0.5 s, were asked to press a key whenever they heard the vowels [a] or [i:]. Reverse-correlation was used to average the spectrum of the noise prior to each key press, thus

estimating the features of the vowels for which the participants were listening. The formant frequencies of these reverse-correlated vowels were similar to those of their respective whispered vowels. The success of this response-triggered technique suggests that it may prove useful for estimating other internal representations, including perceptual phenomena like tinnitus. References: Gold, J., Bennett, P. J., and Sekuler, A. B. (1999). "Identification of band-pass filtered faces and letters by human and ideal observers," Vis. Res. **39**(21), 3537–3560.

1aPP21. Discrimination of rippled spectra in background of maskers of different frequencies. Alexander Supin, Dmitry Nechaev, and Olga Milekhina (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru)

Discrimination of rippled-spectrum signals in masking noise was investigated in normal listeners. The rippled-spectrum signal was 0.5-oct wide centered at 2 kHz. Simultaneous masker was a 0.5-oct wide noise centered below, on, or above the signal band (low-, on-, and high-frequency maskers, respectively). Discrimination of the signals was assessed by (i) threshold for discrimination of ripple spacing; (ii) threshold for discrimination of spectrum-pattern shift. The threshold dependence on the masker level was qualitatively different for the low- and on-frequency maskers. For the on-frequency masker, the masking effect primarily depended on the masker/signal ratio: the thresholds steeply increased at a ratio of 5 dB; no ripple pattern was discriminated at a ratio of 10 dB or higher. Alternatively, for the low-frequency masker, the masking effect primarily depended on the masker SPL: the threshold increased at a masker SPL of 70–80 dB; no ripple pattern was discriminated at a masker SPL of 100 dB. The high-frequency masker produced little effect. Hypothetically, the effect of the on-frequency masker appeared due to a decrease of ripple depth when the masker overlapped the signal, whereas the effect of the low-frequency masker appeared due to widening of the auditory filters at high sound levels.

1aPP22. Relationship of working memory to the effect of listener training on pitch and rhythm perception. Sandra J. Guzman, Julian Milone, Kenneth Meinke (Audio Arts and Acoust., Columbia College, 600 S. Michigan Ave., Chicago, IL 60605, sguzman@colum.edu), Valeriy Shafiro, and Stanley Sheft (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL)

Experience and training can affect discrimination of tonal sequences. Recent work has demonstrated differences in processing multidimensional tonal sequences defined both by pitch contour and sequence rhythm. With groups defined by skill level on a pattern reconstruction task, a significant interaction between group and sequence-task condition suggested a difference in the manner of integration of pitch and rhythm information, with the poorer performing group showing greater integration across dimensions. Current work investigated the effect of training over sessions on pitch and rhythm processing of four-tone sequences by trained musicians and audio-arts students experienced in critical listening. Sequence tones either had a fixed duration (212 ms) with frequency randomly selected from a logarithmically scaled distribution (400–1750 Hz), a fixed frequency (837 Hz) with a randomly selected logarithmically scaled duration (75–600 ms), or a random frequency and duration. In initial conditions, the task was to assemble sequence elements to recreate the target sequence, whereas the later conditions required subjects to ignore extraneous changes in one dimension while reconstructing the target dimension. Trends suggest that training improves performance. Ongoing work is examining the relationship between performance on a number-letter sequencing task and integration of the pitch and rhythm information in tonal sequences.

1aPP23. Contributions of individual frequency bands to the loudness of broadband sounds assessed by loudness matching. Walt Jesteadt, Katyarina E. Brunette, Sara M. Walker, and Katie Thrailkill (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, walt.jesteadt@boystown.org)

Data obtained in sample-discrimination tasks indicate that subjects place high weight on the lowest and highest frequencies when making judgments concerning the loudness of broadband stimuli. In an effort to determine whether these judgments are truly based on loudness, six subjects with normal hearing were tested in a loudness matching task where the stimuli consisted of 15 bands of noise, each two critical bands wide, or 15 tones at the center frequencies of those noise bands. In addition to comparing all combinations of 15-component noise and 15-component tone stimuli, subjects adjusted the level of 15-component stimuli to equate them in loudness with stimuli missing the lowest, middle or highest component. This was done for stimuli with equal levels for each component and for levels approximating the long term average speech spectrum. Noise stimuli had to be increased by 2.5 dB to be equal in loudness to tonal stimuli. Full bandwidth stimuli had to be decreased in level by 1.4–1.8 dB to be equal in loudness to stimuli missing the highest frequency band or tone. The effect was smaller if the middle component was missing and near zero if the lowest component was missing. [Work supported by NIH.]

1aPP24. Binaural pitch averaging and dominance trends in cochlear implant users. Yonghee Oh and Lina A. Reiss (Otolaryngology-Head and Neck Surgery, Oregon Health & Sci. Univ., Mailcode NRC04, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, oyo@ohsu.edu)

Both cochlear implant (CI) and hearing aid (HA) users exhibit broad binaural pitch fusion, the fusion of stimuli differing in pitch across ears by as much as 3 octaves (Reiss *et al.*, 2014, ARO 2015). Typically, two distinct trends are observed for the fused binaural pitch: dominance by the pitch perceived in one ear, or averaging of the pitches perceived in the two ears. In this study, detailed fusion pitch weighting trends as a function of pitch difference were investigated in both bimodal and bilateral CI users. Weights depended on the pitch differences, with averaging occurring for tones or electrodes closer in pitch and dominance occurring for tones or electrodes farther apart in pitch. The averaging region was typically 0.3–0.8 octaves for bimodal CI users and 1–3 electrodes for bilateral CI users around the reference, with weighting bias, in some cases, toward the ear with greater stimulus variability. Overall results show that the fusion pitch trends of the CI users were similar to those observed previously in HA users (Oh and Reiss, submitted); however, CI users showed greater inter-subject variability in both pitch weighting ranges and weighting biases. These findings suggest that binaural pitch averaging could be a potential underlying mechanism of binaural interference in hearing-impaired listeners. [Research supported by NIH-NIDCD grant R01 DC01337.]

1aPP25. Ultrasonic binaural echo perception of object's texture by human echolocation. Miwa Sumiya, Yuki Sarumaru, Taito Banda (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, miwa1804@gmail.com), Kaoru Ashihara (Human Informatics Res. Inst., National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Japan), Kohta I. Kobayasi, Yoshiaki Watanabe, and Shizuko Hiryu (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Echolocating bats use frequency-modulated (FM) and/or constant-frequency (CF) sound for ultrasonic sensing depending on the situation during flight. We investigated discrimination ability of object's texture for sighted subjects to understand acoustic clues for texture recognition in human echolocation. FM and CF ultrasonic echoes from six objects with different materials and surface structures were acquired by a 1/7-size miniature dummy head for presentation of 1/7-times pitch converted binaural audible sounds to listeners through headphones. In the results, averaged rate of correct answer in the case of extremely different surface condition (i.e., acrylic board versus artificial grass) was more than 90% while one in the slightly different surface condition (i.e., acrylic board versus foamed polystyrene) was under 40%. Furthermore, the rate of correct answers in the CF sound condition was approximately 13% lower than one in the FM sound condition. The correlation diagram among targets by multidimensional scaling was dispersed more remarkably in the FM sound condition. When the target pair had slightly different surface condition, differences in the notch pattern of amplitude spectra were observed, especially in the FM sound condition. These suggest that FM ultrasonic binaural sound is more effective for slight-different texture perception than CF ultrasonic binaural sound.

1aPP26. Binaural-cue weighting in free-field localization of narrowband noises presented with open-fit hearing aids and in simulated reverberation. Anna C. Diedesch and G. C. Stecker (Hearing & Speech Sci., Vanderbilt Univ., 7012 Sonya Dr., Nashville, TN 37209, anna.c.diedesch@vanderbilt.edu)

Interaural time (ITD) and level differences (ILD) are susceptible to distortion by multipath acoustics due to reverberation, echoes, and potentially with open-fit, behind-the-ear (BTE) hearing aids, which pose an additional delay between acoustic and processed sound (~2–5 ms). Here, young, normal hearing listeners localized narrow bands of noise centered at 500 or 4000 Hz. Listeners were fit with linear amplification and evaluated in three aided conditions: unaided, open-fit, and occluded BTE coupling. Target sounds were presented over loudspeakers spanning ± 610 azimuth, in anechoic and simulated room conditions. Performance was assessed across conditions by measuring localization gain (slope), variability (R^2), accuracy, and front-back error rate. Results show greater variability in the simulated room than anechoic conditions, particularly for aided conditions (occluded $>$ open $>$ unaided). Aided listening also flattened localization gain, particularly at 4000 Hz. Findings are consistent with acoustic recordings that show reduced ILD and erratic ITD in simulated rooms. The results can be quantified in the form of binaural-cue weighting (ITD/ILD "trading ratio") on the basis of cue values extracted from in-ear recordings obtained for each listener and condition (Diedesch and Stecker 2016, *Assoc. Res. Otolaryngol. Abs.* **39**). [Work supported by NIH R01-DC011548.]

1aPP27. Interactions between front-back confusion and visual capture in summing localization. Christopher Montagne and Yi Zhou (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave., Coor 3430, Tempe, AZ 85283, cmontagn@asu.edu)

Ambiguity in binaural timing and level information often causes front-back confusions in sound localization. This experiment investigated the extent to which front-back confusions are modulated by concurrent visual stimuli. 15-ms duration noise stimuli were presented over two loudspeakers positioned at $\pm 45^o$ in front or behind a listener with a delay between them of -1—1 ms in steps of 0.5 ms, thus evoking summing localization. These sound stimuli were presented with or without light-flash stimuli at three different frontal locations. In comparison to audio only trials, when visual stimuli were also presented front-back confusion decreased for sound stimuli presented in front and increased for rear-presented auditory stimuli. This effect was greater for phantom sound sources than single speaker controls. These results suggest that visual cues influence the perceived location of a sound source when it is outside the field of vision. Further, the phantom sound sources presented using "summing localization" may offer less robust localization cues than physical sound sources, resulting in greater front-back confusion and stronger visual capture. Acoustic analyses of binaural and spectral cues are conducted to identify potential causes of the differences in FBC rates between single speaker and phantom sound sources.

1aPP28. Temporal structure processing reflected by mismatch negativity. T. C. Zhao and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu)

Temporal information comprises significant aspects of complex sounds and is hierarchically organized. Using music, we examined higher-level temporal structure processing. While the note-to-note intervals are variable, the underlying isochronous beats can be realized reliably. Temporal structure processing (i.e., meter) involves further grouping of the beats. Yet, the effects of tempo (beats/min) and meter type have not been examined systematically. In this experiment, electrophysiological methods were adopted to examine how duple and triple meters are processed in different tempos pre-attentively. Using an oddball paradigm, temporal structure (Duple versus Triple), established through alternating strong and weak tones, was violated occasionally (16%). The tempo varied across conditions (Easy versus Hard). The neural sensitivity to this violation is quantified by the magnitude of mismatch negativity (MMN), using EEG. Repeated ANOVA 2 (tempo: Easy versus Hard) x 2 (structure: Duple versus Triple) examined the effects on data from 17 adults (7 males, mean age = 27.07) and results showed that the MMN was significantly larger in Duple than Triple in the Easy condition, while equal between meters in the Hard condition. The results suggest that in an easier tempo, duple structure is better tracked than triple. However, this duple-meter advantage is overridden in difficult tempos.

1aPP29. Additivity of masking at the cocktail party. Gabrielle R. Merchant, Richard L. Freyman, and Karen S. Helfer (Dept. of Commun. Disord., Univ. of Massachusetts Amherst, 358 North Pleasant St., Amherst, MA 01003, gmerchant@umass.edu)

Multi-talker listening environments, like crowded restaurants, are highly complex. In addition to the target talker, the auditory scene may consist of a combination of one or two interfering talkers close to a listener that would likely be relatively high in level (e.g., from people at the same or a nearby table) and numerous additional masking talkers at greater distances whose outputs reach the listener at lower levels. Our understanding of how these masking components combine is limited by the fact that most studies of multi-talker listening present all interfering talkers at the same level. The current study explored the extent to which error rates in a speech understanding task could be predicted from the error rates for two individual components of the masker, specifically from (a) two higher-level (closer) talkers and (b) four lower-level (more distant) talkers. The variables of angular location and level difference of the "closer" and "farther" talkers were explored. At least for an initial subset of conditions in young, normal-hearing listeners, simple additivity of error rates for the individual components appears to be a reasonable model to predict error rates for the combined maskers.

1aPP30. Extent of lateralization of click trains elicited by interaural time differences—A comparison of measurement paradigms. Regina M. Baumgaertel (Dept. of Medical Phys. and Acoust. and Cluster of Excellence 'Hearing4All', Oldenburg Univ., Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, regina.baumgaertel@uni-oldenburg.de) and Mathias Dietz (National Ctr. for Audiol., School of Commun. Sci. and Disord., Western Univ., London, ON, Canada)

Extent of lateralization measurements have previously been performed for various stimulus types, mainly using two measurement paradigms: (1) an acoustic pointer paradigm where the interaural level difference of a pointer stimulus is adjusted to match the intracranial position of a target stimulus and (2) a visual pointer paradigm where the intracranial position of a target stimulus is indicated on a linear scale. Both measurement paradigms are widely used, but rarely within the same study. In this study, the extent of lateralization elicited by unfiltered as well as 4 kHz bandpass-filtered click trains carrying interaural time differences (ITDs) up to 3 ms was measured using an acoustic as well as a visual pointer paradigm. Data from six young normal-hearing listeners were compared. Using the visual pointer paradigm, the extent of lateralization behavior as a function of stimulus ITD for unfiltered as well as bandpass filtered click trains coincided. The extent of lateralization as determined by the acoustic pointer paradigm, however, showed systematic differences at ITDs between 0.2 ms and 1 ms: filtered click trains were matched to smaller pointer ILDs than unfiltered click trains. Caveats and usability of both measurement paradigms will be discussed.

1aPP31. Spatial release from masking in listeners with asymmetric hearing thresholds. Richard L. Freyman, Derina S. Boothroyd, and Decia A. DeMaio (Dept. of Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003, rlf@comdis.umass.edu)

Spatial release from masking was evaluated as a function of target-masker spatial separation in listeners with normal hearing and with chronic asymmetric hearing loss. The target was one of 20 vocoded words spoken by a female talker. The masker was a stream of vocoded nonsense sentences spoken by two other female talkers. Subjects listened in an anechoic chamber with loudspeakers at 1.9 m distance. The target was always presented from directly in front while the masker was presented from the front or from left or right at varying angular separations. In some conditions a copy of the masker with a delay of 4 ms was introduced at the front loudspeaker, simulating a reflection. Subjects detected the target in an adaptive 4AFC task. Results showed strong indications of informational masking in the co-located condition and spatial release that increased with masker angular separation, reaching almost 27 dB at wide separations in normal hearing listeners, and 22 dB even with the reflection. Listeners with asymmetric

thresholds showed high across-listener variability in the spatial conditions but often smaller and strongly asymmetric release, presumably due to head shadow. The reflection reduced this head shadow advantage, and in one subject almost completely obliterated spatial release.

1aPP32. Interaural time difference sensitivity with high rate electrical pulse trains in bilateral cochlear implant users. Alan Kan (Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, ahkan@waisman.wisc.edu)

Interaural time differences (ITDs) are an important cue for obtaining binaural hearing benefits. Bilateral cochlear implant (BiCI) users have shown poor ITD sensitivity when using clinical processors, but have some ITD sensitivity with carefully controlled stimuli presented using synchronized research processors. In BiCI users, low stimulation rates yield best ITD sensitivity, but declines as the stimulation rate increases. However, high rates are typically needed for good speech understanding, which poses a conundrum for maximizing the benefit of BiCIs. Aperiodic high-rate stimulation has been shown to promote ITD sensitivity. However, it is uncertain whether this improvement is due to changes in the loudness level, which may provide an envelope ITD cue. In this study, we tested this hypothesis by measuring ITD just noticeable differences (JNDs) with 4000-Hz electrical pulse trains. Pulse trains either had: (1) constant amplitude and rate; (2) aperiodic timing in the pulses; and (3) aperiodic timing + a random amplitude applied to each pulse. Results have implications in determining whether aperiodic high-rate stimulation can be useful for restoring ITD sensitivity. [Work supported by NIH-NIDCD (R01DC003083 to Ruth Litovsky) and NIH-NICHD (P30HD03352 to the Waisman Center).]

1aPP33. A binaural model using equalization/cancellation and simulated head movements to localize and extract one speaker from a mixture. Nikhil Deshpande and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

This model takes a mixture of two simultaneous speech signals at unique azimuth positions to extract either speaker using the equalization/cancelation (EC) method. Head-related transfer functions are used to spatialize the two sound sources. The model localizes the sources by analyzing interaural time differences and then virtually rotates its head to find the position for the best signal-to-noise ratio. Next, the model segments the mixed speech signal in time and frequency bins, and uses an EC algorithm in each bin to compensate the target signal from the mixture. From the residual non-cancelled energy, it generates a binary map and overlays this on the spectrogram. The ability of the model to cancel out the target signal determines the bins where the target is actually present. The signal is then reconstructed in time and frequency, leaving only one desired target signal. The model achieves signal-to-noise ratios of up to 80 dB. [This material is based upon work supported by the National Science Foundation under Grant Nos. 1320059 and 1539276.]

1aPP34. Method and application of auditory-visual attention training. Andrzej Czyzewski (Multimedia Systems Dept., Gdansk Univ. of Technol., Narutowicza, 11/12, Gdansk, Pomorskie 80-233, Poland, ac@pg.gda.pl), Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland), and Adam Gorski (Multimedia Systems Dept., Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland)

The main idea underlying the proposed attention training is to perform stimulation of the hearing and sight senses employing digital signal processing algorithms controlled by electroencephalography signals. The auditory and visual stimuli are designated to force the perception through hearing and sight senses by the appropriate hemisphere. The applied speech modification uses a non-uniform real-time speech stretching algorithm. The video content retrieval showing the speaker's face is slowed down, accordingly. Research experiments employed subjects with central auditory and visual processing disorders revealing severe communication difficulties. The effectiveness of the proposed method has been shown using formal attention focus tests. It was demonstrated that the proposed method of attention training helps improve speech understanding and reading skills in examined subjects. [Research sponsored by the Polish National Science Centre, Dec. No. DEC-2014/15/B/ST7/04724.]

1aPP35. The trade-off relationship between spatial tuning and multipulse integration in human subjects with cochlear implants. Ning Zhou (Commun. Sci. and Disord., East Carolina Univ., 3310Y Health Sci. Bldg., Greenville, NC 27834, ningzhou1979@gmail.com)

Guinea pigs studies have shown a correlation between the slope of the function relating psychophysical detection thresholds to pulse rate (known as multipulse integration) and the spiral ganglion cell density in electrical stimulation. If this relationship holds true in human subjects with cochlear implants, we hypothesize that multipulse integration should reduce (i.e., become shallow sloped) at stimulation sites with sharp tuning or when electrode configuration changes from broad (monopolar) to focused stimulation (bipolar). Multipulse integration was measured at all stimulation sites using two pulse rates with phase duration of 75 us in a monopolar stimulation mode (MP1 + 2) and bipolar stimulation mode (BP + 0). At selected stimulation sites, a psychophysical spatial tuning curve was also measured using a forward-masking paradigm. The multipulse integration slopes were steeper with MP electrode configuration than with BP. In MP mode, the multipulse integration slopes were steeper at stimulation sites with broader tuning. These results are consistent with the idea that multipulse integration is dependent on the number of available neurons. The degree to which the slopes were affected by focused stimulation, which measured local neural survival near the stimulation sites, was highly correlated with the subjects' speech recognition performance.

1aPP36. Identification and localization of concurrent speech signals in an auditory dual task. Nandini Iyer, Eric R. Thompson, and Brian D. Simpson (Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Wpafb, OH 45433, Nandini.Iyer.2@us.af.mil)

In Air Force operations, there is often a competing demand for auditory attention: listeners must identify and repeat a critical message, while also detecting and localizing the presence of a threat. Previous research (Iyer *et al.*, 2014) has shown significant dual task interference, so that the ability of listeners to maintain a high level of performance on a primary task (keyword identification) was offset by decreased performance in a secondary task (critical call sign detection), relative to the respective single-task performances. The current experiment was designed to examine dual-task performance when the secondary task was changed from detection to localization. Listeners identified color-number keywords originating at 0^o azimuth (primary task), while they localized the presence of a critical call sign originating from one of thirty-two locations surrounding them in 360° on the horizontal plane. The difficulty of the primary task was varied by employing an N-back task (0, 1, or 2) to increase memory load. Results indicated that the cost of dividing attention between the two tasks increased as the memory load in the primary task increased from 1-back to 2-back. More interestingly, while localization accuracy was unaffected, listeners' ability to recognize the critical call sign degraded significantly as the difficulty of the primary task increased. The results have important implications for spatial auditory displays in multitalker environments.

1aPP37. Enhanced auditory spatial performance using individualized head-related transfer functions: An event-related potential study. Matthew G. Wisniewski, Stephanie M. Kenzig, Griffin D. Romigh, Eric R. Thompson, Nandini Iyer, Brian D. Simpson, and Clayton Rothwell (U.S. Air Force Res. Lab., Bldg. 441, Area B, Wright Patterson Air Force Base, OH 45433, matt.g.wisniewski@gmail.com)

Head-related transfer functions (HRTFs) capture location dependent alterations to a sound caused by a physical interaction with a listener's head, shoulders, and outer ears. Sounds rendered with a listener's own HRTF can be localized with accuracy comparable to real sources; however, because anthropometric features vary across listeners, rendering sounds with non-individualized HRTFs can severely degrade localization of virtual sound source location, especially in elevation. Here, we were interested in identifying event-related potential (ERP) features associated with the benefit of employing individualized HRTFs. Thirty "runs" of 250 ms white-noise bursts (6–12 bursts long) with identical virtual elevations (0°–90°, 10° increments) were presented back-to-back within a block. Listeners were instructed to press a button whenever the elevation changed (i.e., at the start of each run). Elevation change detection was enhanced in an Individualized compared to a non-Individualized HRTF condition. ERPs to first bursts of a run showed larger amplitude P3s in the Individualized HRTF condition. Differences in P3 amplitudes between conditions were positively correlated with behavioral benefits gained from employing an individualized HRTF. That effects occurred for the P3 suggests that at least part of the benefit obtained from using individualized HRTFs reflects post-sensory processes.

1aPP38. A functional measurement of auditory spatial acuity through aurally aided visual search. Clayton D. Rothwell (Infoscitex Corp., 4027 Colonel Glenn Hwy, Ste. 210, Dayton, OH 45431, crothwell@infoscitex.com), Brian D. Simpson, and Griffin D. Romigh (Human Effectiveness Directorate, Air Force Res. Lab., Dayton, OH)

Aurally aided visual search (AAVS) uses a spatial audio cue to the location of a visual target in a visual search task (e.g., Perrot *et al.*, 1991), which has shown a reduction in search times when an audio cue is present compared to visual only search. In addition, AAVS has been used to discriminate between the effectiveness of different auditory cues, e.g., free field and virtual (Bolia *et al.*, 1999). This research used an AAVS paradigm in a small frontal region ($\pm 27^\circ$ az $\pm 16^\circ$ el) with high spatial resolution. Nine participants searched for and identified a "C" with a 0.13° opening on either the left or right among distractor "O"s. Crucially, the visual identification was only possible when the target was in the fovea. The visual set sizes used were: 1 (target only), 6, 12, and 24. Participants experienced three conditions: visual only search, virtual audio with individualized head-related transfer functions (HRTFs), and KEMAR HRTFs. Auditory stimuli were 250-ms bursts of broadband noise. Auditory cues speeded search but individualized and KEMAR HRTFs were not different. Further analysis showed that eccentric elevations tended to be slower with KEMAR. Plans to develop this technique are discussed.

1aPP39. Native and non-native intuitions on the phonology of English and French binomials. Viola Green (Dept. of French and Italian, Univ. of Texas at Austin, 201 W 21st St., Stop B7600, Austin, TX 78712-1800, violamakarova@hotmail.com)

Binomial locutions are a well-known case of structural iconicity. By binomial locutions we understand formations that have the shape of A conjunction B (1a), or A-B (1b): (1). a. English: *bread and butter, wear and tear*; French: *dire et juger, ni foi ni loi* b. English: *wishy-washy, helter-skelter*; French: *pêle-mêle, clopin-clopant* The word order in such locutions has been claimed to be motivated by certain constraints on the ordering of the two elements (Cooper and Ross 1975, Birdsong 1995). The main objective of this study is to investigate several specific phonological segmental constraints and the sensitivities that native and non-native speakers of English and French exhibit for these constraints. This is proposed to be tested with a computer-based judgment task, with pairs of nonsensical expressions, structured in such a way that one expression obeys a specific constraint, and the other expression disobeys it. The task of the participants is to listen to such pairs and to indicate which of them they like better by using a 6-point scale. According to the preliminary results, native English speakers tend to be sensitive to most of the phonological constraints in various degrees, while native French speakers are indifferent to most constraints tested.

1aPP40. Rapid reduction of listening effort resulting from predicting speech processing, and delays associated with cochlear implantation. Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, mwinn83@gmail.com)

Speech communication involves not only the recognition of words, but also processing and prediction resulting from those words. This study explored the question of whether prediction guided by semantic context reduces physiological listening effort, and whether this process is as quick and effective in people who use cochlear implants (CIs). To examine this issue, pupil dilation was used as a time-varying index of effort during sentence perception by CI users and listeners with normal hearing. NH listeners also heard spectrally degraded vocoded versions of the stimuli, which consisted of high-and low-context sentences. For NH listeners, context resulted in rapid effort reduction for normal speech; predictable sentences yielded reduced responses even before stimulus offset. For degraded stimuli, this effect was not observed until after the stimulus was over, suggesting that listeners normally process and take benefit from context in real time, but poor signal quality delays that process. For CI listeners, effort reduction from context was generally not as early or as large as that for NH listeners. These effects all persisted even when intelligibility was perfect, suggesting that word and sentence recognition scores could mask difficulties in functional language use, like timing and prediction in speech comprehension.

1aPP41. Using sociolinguistic phonetic perception to fine tune cochlear implant simulations. Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, mwinn83@gmail.com)

Simulations of the sound of cochlear implants (CIs) are used for people with acoustic hearing for counseling, pedagogy, and to rigorously control stimulus parameters in predictions of CI performance. The most commonly used CI simulation is the eight-channel noise vocoder, which approximates intelligibility scores in better-performing CI users. However, this method underestimates the ability of CI users to demonstrate subtle effects in phonetic perception, including the adjustment of phonetic perceptions based on talker gender (phonetic "accommodation"), which CI users can do, despite no such effect found in normal-hearing listeners when using the eight-channel noise vocoder. Toward the goal of simulating CI performance more accurately, gender-driven phonetic accommodation was measured using three styles of vocoder, including an all-channel (conventional style) noise vocoder, and a peak-picking noise vocoder that delivers the top eight channels out of 22 per time bin. Carriers were variable-bandwidth noise or a pulsatile harmonic complexes, each delivered with multiple levels of spectral resolution. CI-like phonetic accommodation effects emerged in varying degree for all vocoders except the conventional eight-channel noise vocoder, suggesting that, despite good match of overall intelligibility, it is limited in its ability to predict more subtle aspects of speech perception.

1aPP42. Consonance preferences are not universal: Indifference to dissonance among native Amazonians. Josh McDermott (Brain and Cognit. Sci., MIT, 77 Massachusetts Ave., 46-4065, Cambridge, MA 02139, jhm@mit.edu), Alan Schultz (Anthropology, Baylor Univ., Waco, TX), Eduardo Undurraga, and Ricardo Godoy (Heller School for Social Policy and Management, Brandeis Univ., Waltham, MA)

In Western cultures, some combinations of musical notes are consonant (pleasant) and others are dissonant (unpleasant). Although the aesthetic contrast between consonance and dissonance is commonly thought to have biological roots, and thus to be universally present in humans, a definitive test has remained elusive. Here we report experiments with the Tsimane'—a native Amazonian society with minimal exposure to Western culture—and comparison populations elsewhere in Bolivia and the USA that varied in exposure to Western music. Participants rated the pleasantness of sounds. Despite exhibiting Western-like preferences for several other aesthetic contrasts, and Western-like discrimination abilities, the Tsimane' rated consonant and dissonant chords and vocal harmonies as equally pleasant. In contrast, Bolivian city- and town-dwellers exhibited significant preferences for consonance, albeit to a lesser degree than U.S. residents. The results indicate that consonance preferences dramatically vary across cultures, and can be fully absent in some cases. The observed variation in preferences is plausibly determined by exposure to Western music, suggesting that culture plays a dominant role in shaping aesthetic responses to consonance.

1aPP43. Difference between Mandarin Chinese and English speech masking caused by signal-driven processes. Jing Chen, Hongying Yang, Zilong Xie, Liang Li, and Xihong Wu (Speech and Hearing Reseach Ctr., Ctr. for information Sci. Peking Univ., Rm. 2228, No. 2 Sci. Bldg., Beijing 100871, China, chenj@cis.pku.edu.cn)

Speech intelligibility in competing speech is worse in native masking than in non-native masking, which could be caused by signal-driven processes and by knowledge-driven processes. To evaluate the contribution of signal-driven processes, the speech intelligibility of Mandarin Chinese was measured on native Chinese listeners in four types of maskers for both Mandarin and English masking: (1) 1-talker's normal speech (NOR); (2) amplitude-modulated noise with speech envelope (AM); (3) speech-like signals synthesized according to the speech F0 modulation with steady temporal envelope (FM); (4) speech-like signals with speech envelope (a combination of AM and FM, AFM). The results show that speech reception threshold in Chinese masking was about 4 dB higher than that in English, in which about 2 dB was caused by the AM, −1 dB was caused by FM, and 1 dB was caused by AFM. The results revealed the signal-driven processes might play minor role in the difference of speech masking between Mandarin Chinese and English. The effects of amplitude modulation and F0 fluctuation were analyzed and discussed.

1aPP44. Speech recognition performance of listeners with normal hearing, sensorineural hearing loss, and sensorineural hearing loss and bothersome tinnitus when using air and bone conduction communication headsets. Candice A. Manning (US Army Res. Lab., U.S. Army, 3710 SW US Veterans Hospital Rd., P5, Portland, OR 97239, Candice.Manning@va.gov), Timothy J. Mermagen, and Angelique A. Scharine (US Army Res. Lab., U.S. Army, Aberdeen Proving Ground, MD)

Military personnel are at risk for hearing loss due to noise exposure during deployment (Norin *et al.*, 2011). Despite mandated use of hearing protection, hearing loss and tinnitus are prevalent due to reluctance to use hearing protection. Bone conduction (BC) headsets can offer good speech intelligibility for normal hearing (NH) listeners while allowing the ears to remain open in quiet environments and the use of hearing protection when needed. Tinnitus sufferers often show degraded speech recognition; however, it is unclear whether this is a result of decreased hearing sensitivity or increased distractibility. It has been suggested that vibratory stimulation of BC might ameliorate tinnitus; however, there is currently no research to support or refute this claim (Hoare, *et al.*, 2014). Speech recognition of words presented over AC and BC headsets was measured for three groups of listeners: NH, sensorineural hearing impaired, and/or tinnitus sufferers. Three levels of speech-to-noise ($SNR = 0, -6, -12$) were created by embedding speech items in pink noise. Better speech recognition performance was observed with the BC headset regardless of hearing profile and speech intelligibility was a function of SNR. Discussion will include study limitations and the implications of these findings for those serving in the military.

1aPP45. Evaluation of frequency-dependent hearing loss by words identification. Jing Chen, Hongying Yang, and Xihong Wu (Speech and Hearing Res. Ctr., Key Lab. of Machine Percept. (Ministry of Education), Dept. of Intelligence Sci., Ctr. for information Sci., Peking Univ., Sci. Bldg., Rm. 2228, No. 2, Beijing 100871, China, chenj@cis.pku.edu.cn)

Pure-tone audiometric test can reflect decrease of auditory sensibility at certain frequencies for the hearing loss; however, it requires critical environment and not suitable for the self-test in general environment, e.g., for some related applications on smart equipments. This study was aimed to evaluate the frequency-dependent hearing loss by words identification tests. Thirty-nine word-pairs in Mandarin Chinese were selected according the vowels/consonants confusion matrix. Every word is a monosyllable with CV (consonant-vowel) structure, and in each pair, the two words are easily confused by their formants (the same consonant conjunct with two different vowels) or the formant transitions (two different consonants conjunct with the same vowel). Hearing loss was simulated by band-stop filtering with different center frequencies. Eight filtering conditions, corresponding to eight center frequencies of the band-stop filter, were tested for nine normal hearing young students. The results show the word identification was significantly decreased for one certain filtering condition (worst-performance frequency, WPF) for 16 word-pairs, indicating these word-pairs could be effective to evaluate the frequency range of hearing loss. The relation of the WPF and the formant/formant transition frequencies were analyzed and discussed.

1aPP46. An algorithm to increase speech intelligibility for hearing-impaired listeners in entirely novel noises. Jitong Chen, Yuxuan Wang (Comput. Sci. and Eng., The Ohio State Univ., Columbus, OH), Sarah E. Yoho (Speech & Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, Columbus, OH, yoho.17@osu.edu), DeLiang Wang (Comput. Sci. and Eng., The Ohio State Univ., Columbus, OH), and Eric Healy (Speech & Hearing Sci., The Ohio State Univ., Columbus, OH)

Supervised speech segregation has been recently shown to improve human speech intelligibility in noise, when trained and tested on similar noises. However, a major challenge involves the ability to generalize to entirely novel noises. Such generalization would enable hearing aid and cochlear implant users to improve speech intelligibility in unknown noisy environments. This challenge is addressed in the current study through large-scale training. Specifically, a deep neural network (DNN) was trained on 10,000 noises to estimate the ideal ratio mask, and then employed to separate sentences from completely new noises (cafeteria and babble) at several signal-to-noise ratios (SNRs). Although the DNN was trained at the fixed SNR of 2 dB, testing using hearing-impaired listeners demonstrated that speech intelligibility increased substantially following speech segregation using the novel noises and unmatched SNR conditions of 0 dB and 5 dB. Sentence-intelligibility benefit was also observed for normal-hearing listeners in most noisy conditions. The results indicate that DNN-based supervised speech segregation with large-scale training is a very promising approach for generalization to new acoustic environments.

1aPP47. Contribution of near- and suprathreshold hearing deficits to speech recognition. Anna Warzybok, Sabrina Pieper, and Sarah Verhulst (Medical Phys. and Cluster of Excellence Hearing4all, Univ. of Oldenburg, Universität Oldenburg, Oldenburg D-26111, Germany, a.warzybok@uni-oldenburg.de)

This study addresses the contribution of audibility and suprathreshold deficits to speech recognition in quiet, stationary, and speech-modulated noise. The relative importance of these deficits for different frequency ranges of hearing was determined by measuring speech recognition using broadband speech and noise, but also high- and low-pass filtered versions of these stimuli. Audibility was assessed using pure tone thresholds and speech reception thresholds (SRT) in quiet. Three listener groups were considered to parse out hearing deficits: young (yNH) and older normal-hearing (oNH) listeners and hearing-impaired (HI) listeners. oNH and HI listeners performed equally poor in stationary noise where yNH listeners showed significantly better SRTs in both noise conditions. Since audibility did not correlate with SRTs in stationary noise but with modulated noise, and SRTs in stationary and modulated noise correlated with each other, a common mechanism (not audibility) must affect speech recognition in noise. Furthermore, our results suggest that performance in the broadband condition cannot be explained by the low frequency mechanisms since all groups performed similar in the low-pas conditions whereas the oNH and oHI group performed equally bad in high-pas condition. Instead, we found evidence for a contribution of high frequency envelope coding mechanisms to broadband performance

1aPP48. Spectral-normalization filter for subjective analysis of the aging voice. Mark L. Berardi, Eric J. Hunter (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Rm. 211D, Oyer Speech & Hearing Bldg., East Lansing, MI 48824, berardi1@msu.edu), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Voice quality changes with age. One way of identifying voice quality changes and the possible physiological degeneration related to vocal function is through perceptually estimating talker age and then correlating estimated talker age with acoustical analysis. The most common perceptual studies investigating estimated talker age are cross-sectional. While these studies are useful, longitudinal studies provide additional details in the progressive degeneration of a single speaker's (or group of speakers') voice quality. Nevertheless, one limitation of longitudinal studies is that perceptual ratings of voice quality or talker age could be biased by recording quality (e.g., the spectral qualities of recordings from earlier decades are limited by the technology used). In this paper, a spectral-normalization filter was developed and applied to a corpus of recordings from an individual spanning about 50 years (1959–2007). The filter was shown to be effective in normalizing the autospectra of the recordings. The fundamental frequency was unaffected by the filter, suggesting that the core characteristics of the voice were unchanged. Preliminary subjective analysis suggests that, in spite of significant differences in the original recording equipment, the recording quality of all the files was perceptually similar.

1aPP49. Attenuation of hearing protectors: A systematic comparison of subjective and objective measurement methods. Hugues Nélisse (IRSST, 505 Blvd. De Maisonneuve Ouest, Montreal, QC H3A 3C2, Canada, hugues.nelisse@irsst.qc.ca), Cécile Le Cocq (Génie Mécanique, École de Techonologie Supérieure, Montreal, QC, Canada), Jérôme Boutin (IRSST, Montreal, QC, Canada), Frédéric Laville, and Jérémie Voix (Génie Mécanique, École de Techonologie Supérieure, Montreal, QC, Canada)

A key component when selecting a hearing protector is the noise attenuation offered by the device. The subjective Real-Ear Attenuation at Threshold (REAT) test method is the most commonly used procedure to measure attenuation. On the other hand, with the increase popularity of individual fit testing and miniaturization of electronic components, the Microphone-In-Real-Ear approach (MIRE), and its field counterpart F-MIRE, are becoming more appealing and well suited for estimating attenuation in laboratory or in "real world" occupational conditions. In this approach, two miniature microphones are used to measure sound pressure levels in the ear canal under the protector and outside of the protector. This study presents a systematic evaluation of the various factors relating the subjective and objective attenuation values. Experiments on human subjects were carried out where the subjects were instrumented on both ears with microphones outside and underneath their protector. They were then asked to go through a series of subjective hearing threshold measurements followed immediately by microphone recordings using high level broadband noises. Earmuffs, earplugs, and dual-protection were tested for each subject. The various factors relating the subjective and objective attenuation data are first presented. Results showing the relative importance of these factors are presented and discussed as well as various comparisons obtained with the different attenuation values.

1aPP50. Contributions to high-frequency reflectance from the change in diameter between sound-delivery tube and ear canal. James D. Lewis and Mary Easterday (Audiol. and Speech Pathol., Univ. of Tennessee Health Sci. Ctr., 578 South Stadium Hall, Knoxville, TN 37921, jdlewis@uthsc.edu)

Measurement of wideband reflectance requires stimuli to be delivered to the ear canal through a tube with diameter seven-times smaller than that of the average canal. This discontinuity may contribute to the measurement-location sensitivity of reflectance above 5 kHz [Lewis and Neely. (2015). J. Acoust. Soc. Am. **138**, 977–993]. To test this hypothesis, reflectance was measured at different locations in a set of artificial ear canals (cylindrical tubes with diameters of 8, 9, and 10 mm) terminated by an IEC711 coupler, and compared to the theoretical reflectance. The measured reflectance exhibited a high-frequency notch that (1) decreased in frequency as distance between the probe and coupler increased and (2) increased in magnitude as canal diameter increased. The theoretical reflectance lacked the high-frequency notch of the measured reflectance; however, it was possible to simulate the measured data by adding an inductance in series with the canal model [Karal. (1953). J. Acoust. Soc. Am. **25**, 327–334]. Transforming the impedance measured in the artificial canals to remove the estimated inductance decreased the notch depth and measurement-location sensitivity of the measured reflectance, especially for the larger diameter canals. Additionally, the measured reflectance more-closely approximated the theoretical reflectance. Findings have implications for *in-situ* sound-level calibrations.

1aPP51. Effect of smartphone orientation on sound level measurement. Niall A. Klyn and Lawrence L. Feth (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, klyn.1@osu.edu)

The proliferation of smartphones provides an opportunity for acoustic measurements to be made by citizen scientists. However, there are concerns about the accuracy of data collected by the public. This study reports on the variability in smartphone sound level measurements due to how the device is being carried. A sample of smartphones running either Android or iOS and with various microphone placements were selected and calibrated for dBA measurements. These devices were tested in anechoic and semi-reverberant rooms while being situated to reflect common carrying positions. We found that the measured sound level in the anechoic chamber could vary by >10 dB depending on the orientation of the device, the location of the microphone, and the relationship of the device to the sound source. Smaller but still substantial variation was observed in the semi-reverberant space. These data highlight the variability probable in participatory sensing efforts, and underscore the potential usefulness of providing guidelines for data collection by citizen scientists. [Work supported by the Battelle Engineering, Technology and Human Affairs Endowment.]

1aPP52. Individual differences in self-adjustment of hearing aid amplification. Trevor T. Perry, Peggy Nelson, Dianne VanTasell, and Melanie Gregan (Hearing, Speech, and Lang. Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, trevortperry@gmail.com)

Listening in noisy environments remains a common complaint among hearing aid users. Self-adjustment of amplification parameters could improve listener satisfaction in noise. Previous data (Nelson *et al.*, AAS 2015) indicated considerable variability among hearing-impaired listeners when self-adjusting amplification in noisy environments, but the sources of variability are not understood. Data will be presented from 30 adult listeners with mild to moderate hearing loss who self-adjusted amplification parameters in laboratory-simulated restaurant environments. Participants listened using a real-time simulation of a multichannel compression hearing aid and adjusted gain/compression parameters via a simple user interface (EarMachine). On average, listeners selected less gain relative to NAL-NL2 targets, and they reduced gain as noise levels increased. Hearing loss did not predict variation in self-selected amplification. Small but significant gender differences were observed. Female subjects selected more high-frequency gain (2000–8000 Hz) but less low-frequency gain than male subjects. Listeners with prior hearing-aid experience selected more high-frequency gain than listeners with no hearing-aid experience. Younger age was associated with selecting more low-frequency gain. Within-group variability was large compared to mean differences between groups and only a small proportion of variance was explained. These data suggest that self-adjustment of amplification was not easily predicted by listener-specific factors.

1aPP53. A study on the effects of depression using the weighting white noise. Seonggeon Bae (Daelim Univ., 29, Imgok-ro, Dongan-gu, Anyang-si, Gyeonggi-do, 431-715, Korea, Anyang 431-715, South Korea), sgbae123@empal.com) and Myungjin Bae (Soongsil Univ., Seoul, South Korea)

In general, depression is a disease that appears negative emotions and changes negative emotions in brain function. Fifteen percentage of the population suffers from anxiety disorders and depression at least once during a lifetime. Delta waves are often found to increase in depressed patients than in the general population, which cause negative emotions. The alpha waves increases in the left frontal lobe, and the beta waves increases in the right frontal lobe. Depending on the brain waves, patients with depression have such characteristics. In this study, we used a white noise for the symptoms of depression. Analysis of depression in a white noise is advantageous than the using EEG features. White noise weight has an effect on depression.

1aPP54. The use of auditory models for the prediction of protected-ear localization. Griffin D. Romigh (Air Force Res. Labs, 2610 Seventh St., Area B, Bldg. 441, Wright Patterson AFB, OH 45433, griffin.romigh@us.af.mil) and Eric R. Thompson (Air Force Res. Labs, Dayton, OH)

Several authors have shown that listeners' natural auditory localization performance is degraded significantly when wearing a hearing protection device (HPD) (Atherley and Noble, 1970; Abel and Armstrong, 1993; Bolia *et al.*, 2001; Brungart *et al.*, 2007; Hobbs *et al.*,2008). Research suggests that this degradation in protected-ear localization caused by the disruption of the natural spatial hearing cues by physical obstructions such as a plug or muff, as well as active systems' artifacts like a limited frequency response, temporal processing delay, or artificial directionality. The current work investigated the utility of using existing auditory localization models from The Auditory Modelling Toolbox (Sondergaard and Majdak (2013)), along with measurements of the device directional transfer function, to accurately predict behavioral protected-ear localization performance. Ten listeners were asked to localize 250-ms broadband noise bursts originating from 245 spatial locations surrounding the listener while wearing each of 11 different HPDs (a mixture of active and passive, plugs and muffs). Directional transfer functions were also collected for each subject, device, and location. Results indicate good agreement between observed behavioral results and model predictions when viewed macroscopically (global mean error values), but less agreement was seen on a trial-by-trial basis.

Session 1pAA

Architectural Acoustics and Speech Communication: Sound System Design, Optimization, and Intelligibility

Peter Mapp, Cochair
Peter Mapp Associates, Copford, Colchester CO6 1LG, United Kingdom

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362

Chair's Introduction—1:00

Invited Papers

1:05

1pAA1. Some observations on potential errors in acoustic and electroacoustic systems computer modeling. Peter Mapp and Josh Boatman (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com)

Although computer modelling of sound systems is now a mature science, it is surprising how many inaccurate models are still being made. The paper identifies and discusses the most common causes for modeling errors that the authors' regularly encounter. These include (1) lack of model detail and dimensional accuracy; (2) use of statistical calculations in non-diffuse or inhomogeneous (statistical) spaces; (3) errors due to modeling of materials with unknown absorption and scattering properties; (4) incorrect (or corrupt) loudspeaker data; (5) incorrect consideration of boundary effects and net energy flows; and (6) use of statistical calculations where discrete, long path echoes and reflections are present. Examples of each of the above error mechanisms will be presented together with their typical resultant calculation errors.

1:25

1pAA2. The importance of phase, attention, intelligibility, and recall in sound system design. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt. Auburn St #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Measures for the accuracy of speech transmission over an audio channel, such as Speech Transmission Index, rely on random word recognition as the standard for quality. In this paper, we propose that the ability to recall information at a later time is a more useful standard for sound quality than simply recognizing words. Recall, particularly in a complex or noisy environment, depends critically on the focused attention of a listener. There is evidence that "proximity," the auditory perception that a source is close to a listener, enhances both focused attention and the ability to localize and separate sound sources from competing signals. We find that proximity and the benefits it brings depend on the phase alignment of frequencies above 1000 Hz. This alignment is lost when there are excess reflections, reverberation, noise, or multiple loudspeakers. We will present measurement techniques that can quantify proximity from an impulse response, and sound designs that maximize it in practice.

1:45

1pAA3. The physics of auditory proximity: Auditory mechanisms that extract pitched signals from noise and other signals. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt. Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Cutthroat evolution has given us seemingly magical abilities to hear speech in complex environments. For example, we can tell instantly, independent of timbre or loudness, if a sound is close to us. In a crowded room, we can switch attention at will between at least three different simultaneous conversations, and involuntarily switch to one of them if our name is spoken. These feats are only possible if, without conscious attention, each voice has been separated into an independent neural stream. The separation process relies on the phase relationships between the harmonics above 1000 Hz that encode speech information, and the neurology of the inner ear that has evolved to detect them. When phase is undisturbed, once in each fundamental period harmonic phases align to create massive peaks in the sound pressure at the fundamental frequency. Pitch-sensitive filters can detect and separate these peaks from each other and from noise with amazing acuity. But reflections and sound systems randomize phases, with serious effects on attention, source separation, and intelligibility. This paper will describe the many ways ears and speech have co-evolved, and recent work on the importance of phase in acoustics and sound design.

2:05

1pAA4. "Voice lift"—Improving sonic perception, including intelligibility, using multi-channel time variant systems. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

For many of our clients, the "optimum" electro-acoustic reinforcement system is one that has a "magic microphone." This translates to something that they do not have to wear or hold, or stand in a specific spot to use. It does not need to have its battery replaced or cable plugged in. They might not even want to "flip a switch" to turn it on—it would just "be there" and all that they would need to do is talk. And if someone in the audience wanted to ask a question, they too would have a "magic microphone" that would enable everyone to hear them. While little has changed with the development of flying cars, advancements in multi-channel digital signal processing, and microphone arrays have made audio systems that function in this manner attainable. This paper will discuss factors that are important for intelligibility, how this differs from factors that are important for the perception sonic quality, and why both are required in order to achieve a comfortable and engaging listening experience. It will also describe the underlying complexity involved in designing and integrating such systems. Two recent examples will be cited—the Best Buy Theatre in Minneapolis, Minnesota, and the Netlflix corporate auditorium in Los Gatos, CA.

2:25

1pAA5. Sound system analysis and design of the Minnesota State House. Bruce C. Olson and Ana M. Jaramillo (Olson Sound Design LLC, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana@olsonsound.com)

During the renovations of the Minnesota Capitol we were called to advise on the sound system of the House of Representatives. In order to assess the quality of the current system we performed noise, RT and STI measurements. Using the collected information we created an EASE model for the room to look at improvement proposals. This presentation addresses the differences between measured and simulated (predicted) STI values and the use of modeling tools for design. Due to the type of client we had for this project (final user, political) we were asked to report findings and recommendations in a presentation format and relied on auralizations as a tool for demonstrating changes in speech intelligibility between the existing and proposed sound systems.

2:45

1pAA6. Listening for solutions to a speech intelligibility problem. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

An historic Temple, built in 1929 in Los Angeles with 1800 seats and a 100-ft diameter dome, recently underwent a $150 million renovation, including a new sound system. Even after several system tunings, complaints of poor speech intelligibility remained. The system designers recommended additional sound absorptive treatment, as apparently based on their measurements, but this would be expensive, and would affect the visual aesthetic and perhaps the quality of live unamplified music. Our firm was requested to visit and offer assistance. Using a click track played over the system and simply listening, it was quickly deduced that turning off some of the loudspeakers would result in significant improvement in speech intelligibility throughout the entire Temple. This paper will discuss various aspects of the background, analysis, and client responses, as well as the historicity of this Temple of the Stars.

3:05–3:15 Break

3:15

1pAA7. Sound system Déjà Vu—A chance to revisit the design approach in an acoustically challenging multi-purpose space. Deb Britton (K2 Audio, 4900 Pearl East Circle, Ste. 201E, Boulder, CO 80301, deb@k2audio.com)

More than a decade ago, we were tasked with designing a sound reinforcement system that would provide great speech intelligibility, in a highly reverberant space, without modifying any of the architectural finishes. The sound system had to support different types of events, held in different locations within the space, and with varying audience sizes. In 2015, we were asked to revisit that original loudspeaker design, and upgrade the sound system in that same space. This paper presents a case study of both upgrades, and describes the challenges, the lessons learned, and the changes in approach from the original upgrade to the most recent sound system design.

3:35

1pAA8. A presentation on the Infocomm spectral balance standard. John A. Murray (Standards, InfoComm Int.., 90 Stanford Pl., Woodland Park, CO 80863, john@optimumss.com)

InfoComm is finalizing a Spectral Balance Standard soon to be released for public review. The standard is not a "how to" for system equalization; it only verifies properly equalized systems. It defines a validation process that ensures sound systems reproduce a relatively uniform spectral balance for (3) classes of systems: full-bandwidth, limited bandwidth, and voice-communications. As conformance to the Standard confirms optimum spectral performance, this makes it an integral part of a complete optimization suite. And, as a part of system optimization, this makes the Spectral Balance Standard relevant to the Sound System Design, Optimization, and Intelligibility ASA Special Session. The measurement portion divides the audio spectrum into two sections: impulse-response measurements above the modal range of each room and steady-state spatially averaged measurements in and possibly below the modal range. The two measurement types are conjoined for each measurement position and a range of variation limits and fractional-octave smoothing criteria, specific to each class of sound system, are applied. Sound system responses that fit within the limits conform to the standard. This presentation will discuss the overall philosophical approach towards a spectrally balanced sound system that drives the specifics of the measurements, limits, and criteria chosen for the Standard.

3:55

1pAA9. Generalized energy density and its significance for sound systems in rooms. Timothy W. Leishman and Zachary R. Jensen (Phys. and Astronomy, Brigham Young Univ., N247 ESC, Provo, UT 84602, twleishman@byu.edu)

The room constant has long been used for fundamental architectural acoustics predictions, including those related to sound systems. It is an integral part of the Hopkins-Stryker equation, which characterizes semi-reverberant fields via total energy density. While many have used potential energy density instead of total energy density with the equation, modern measurement techniques have facilitated the use of the latter. Recently, generalized energy density has been advanced as a metric with even greater spatial uniformity in reverberant fields and attendant benefits. This presentation will review the quantity and discuss how it may be properly used with the Hopkins-Stryker equation. It will also explain how the quantity may improve direct measurement of the room constant rather than requiring its estimation from approximate reverberation times, room geometrical values, and absorption coefficients. Generalized energy density may likewise be used in conjunction with impulse response measurements, which are beneficial in other ways for sound system measurements and optimizations.

4:15

1pAA10. A review of sound reinforcement system design and applications as continually changing over the years through the introduction of new design techniques, equipment, and commissioning. Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd, Lawrence, KS 66045, coffeen@ku.edu)

Since the author's first commercial sound system design in 1960 and the first use of this system in 1962 system design techniques, audio equipment, and methods used for system test, adjustment, and equalization have been continually changing. Most of these changes have been good and have resulted in better performing sound reinforcement systems. However, a few of the changes have not contributed to improved system performance. This presentation will review the following changes: in design methods such as techniques for determining loudspeaker sound distribution in a particular venue, in audio equipment including the use of digital signal processing and audio signal transmission, in loudspeaker directivity both better and worse, in the introduction of new microphone types and the disappearance of several useful microphone types, and in system commissioning including equalization from the days of passive filters to digitally produced filters. The presentation will include several microphone and equalization demonstrations.

4:55

1pAA11. Distributed mode loudspeakers—An enigma or just a misunderstood phenomenon? Peter Mapp (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com)

Distributed mode loudspeakers (DMLs) have been commercially available for nearly 20 years, but they are still misunderstood and often misapplied in audio and sound reinforcement installations. The paper will discuss the history and basic properties of DMLs and look at current examples and their potential application in commercial audio and sound reinforcement. In particular, often quoted claims concerning their coverage, diffuse radiation, and intelligibility will be discussed together with a discussion of appropriate measurement parameters and measurement techniques that may be used to characterize these devices.

5:15–5:40 Panel Discussion

Session 1pAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Comparative Hearing—Honoring Dick Fay

Arthur N. Popper, Cochair
University of Maryland, Biology/Psychology Bldg., College Park, MD 20742

William Yost, Cochair
ASU, PO Box 870102, Tempe, AZ 85287

Chair's Introduction—1:30

Invited Papers

1:35

1pAB1. Dick Fay, man of many tastes. William Yost (Speech and Hearing Sci., Arizona State Univ., PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Hearing in Vertebrates: A Psychophysics Databook and *Hot Dog Chicago* seem like fitting bookends for the amazing career of Richard (Dick) R. Fay. Both books were researched and written in the 1980s, a fertile time in Dick's career. The *Databook* was written by Dick and *Hot Dog Chicago* with his friend and colleague at Loyola University Chicago, Rich Bowen (Rich is a Professor of Psychology studying visual psychophysics). The *Databook* is unique, representing a scholarly tour-de-force which has become the '"bible" for those interested in vertebrate behavior and hearing; and *Hotdog Chicago* is an informative Chicago fast-food restaurant guide written with a whimsical sense of humor and read by hundreds, if not thousands, interested in an iconic Chicago culture. In the years-long preparation and production of the *Databook*, Dick arrived at a new view of the advantages of the comparative study of auditory processing. This talk will describe Dick's efforts in writing these two books and how those efforts helped shape him as a scholar and friend of so many.

1:55

1pAB2. Dick Fay's contributions to bioacoustics. Arthur N. Popper (Univ. of Maryland, Biology/Psych. Bldg., College Park, MD 20742, apopper@umd.edu)

For well over 40 years, Dick Fay has been making broad, significant, and insightful contributions to our understanding of vertebrate, and particularly fish, hearing. His work has range from psychoacoustics to physiology to, most recently, behavioral responses. While Dick has worked on a number of different species, his primary research animal has been the goldfish (though he has also worked with sharks, cichlids, toadfish, and other species). Through ingenious studies that involved a variety of different psychophysical techniques, Dick not only asked question about what fish can hear, but also did pioneering work on things like masking, critical bands, and auditory scene analysis. Dick's physiological studies have been of equal importance. These often involved the use of an ingenious "shaker table" that he developed which allow very precise control of the stimulus in three-dimensions. Using this approach, Dick has asked questions about function at many levels of the auditory system, providing important insight not only in fish hearing and their potential for sound source localization, but also in the evolution of the vertebrate auditory system.

2:15

1pAB3. Fish hearing and bat sonar. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu)

The Dynamic Duo of Dick Fay and Art Popper advanced auditory research by bringing to light the diversity of hearing capabilities and auditory mechanisms in different species of fishes. I was lucky to have been a graduate student along with Dick in Glen Wever's Auditory Research Laboratory at Princeton—a true bastion of comparative hearing science. Fish auditory mechanisms fit into an evolutionary pattern of auditory capabilities that spans the entire vertebrate family tree. There is a ubiquitous hearing capability that covers frequencies of 20–30 Hz to perhaps 500 Hz that is probably shared by all vertebrates. It is mediated by phasic nerve discharges that convey sound period to higher auditory centers by well-defined synchronization to cycles of sounds in this low-frequency region. This regime is critical for coding many vowel-like animal vocalizations that have first or higher harmonics in the temporal coding region. Although the mammalian cochlea is best known for tonotopic representation—important especially for tone frequencies and wideband sounds above 1–2 kHz, a wide variety of vertebrates in fact have evolved place coding through tonotopic organization of auditory receptors. These two coding regimes operate in parallel for biosonar imaging, too. [Work supported by ONR and Capita Foundation.]

2:35

1pAB4. What is so special about goldfish? Peggy L. Edds-Walton (Education, S.E.E., UC Riverside, Riverside, CA 92506, seewalton@gmail.com)

The goldfish (*Carassius auratus*) was the preferred piscine subject for the auditory research conducted by Dick Fay for many years. As one of the otophysine teleosts, the goldfish has been called an "auditory specialist," and much of the research Fay and his collaborators conducted on goldfish focused on aspects of the sense of hearing (e.g., frequency selectivity and intensity discrimination), using classical conditioning (respiration or heart rate) and a variety of psychophysical procedures to determine what goldfish could hear. Studies of toadfish (*Opsanus tau*) and mottled sculpin (*Cottus bairdii*) provided additional insight on frequency tuning and intensity discrimination in "nonspecialist" teleost fishes *and* unique insights into how the brain of "nonspecialists" encodes source direction, indicating that goldfish are not so special.

2:55

1pAB5. Auditory research in fish: From the goldfish to zebrafish. Zhongmin Lu (Biology, Univ. of Miami, 5151 San Amaro Dr., Cox Annex 208, Coral Gables, FL 33158, zlu@miami.edu)

Dr. Richard R. Fay has made seminal contributions to our understanding of hearing and bioacoustics in fish. I was fortunate to be trained as a Ph.D. student in fish auditory physiology under his guidance. For this presentation, I will first review my early graduate work on auditory processing in goldfish with Dick and express my deepest appreciation and thanks to him for helping me start my postdoctoral training in Dr. Arthur Popper's lab as well as his positive influence on my academic career. For the second part of the talk, I will switch to my current research focus on genetic hearing loss using zebrafish. For the past decade, the zebrafish (*Danio rerio*) has emerged as an important vertebrate model for biomedical research because it combines many advantages such as excellent embryology, exceptional *in vivo* imaging, and powerful genetics in one organism. Collaborating with an otolaryngologist and a clinical geneticist in the University of Miami Miller School of Medicine, my lab has been assessing auditory functions of novel deafness genes. I will show an example of our project on the zebraifsh model for sensorineural hearing loss using molecular, genetic, behavioral, confocal imaging, and electrophysiological approaches.

3:15–3:30 Break

3:30

1pAB6. Frequency tuning and directional preferences of eighth nerve afferents in the non-teleost bony fish, *Acipenser fulvescens*. Michaela Meyer (Biology, Univ. of Maryland, 3 Blackfan Circle, Ctr. for Life Sci., 14th Fl., Rm. 14021, Boston, MA 02115, meyerghose@gmail.com)

We investigated the coding mechanisms for spectral analysis and sound source location in *Acipenser fulvescens*, the lake sturgeon. *A. fulvescens* belongs to one of the few extant non-teleost ray-finned fishes, with a phylogenetic history that dates back about 200 million years. A shaker system was used to simulate the particle motion component of sound during electrophysiological recordings of isolated single units from the eighth nerve. Data were compared to teleosts and land vertebrates. Peripheral coding strategies for spectral analysis and sound source location in *A. fulvescens* resembled those found in teleosts. Frequency characteristics generally resembled that of low-frequency afferents of other fishes and land vertebrates and the auditory periphery in *A. fulvescens* appears to be well suited to encode the intensity of sound. Eighth nerve afferents responded to directional stimuli in a cosine-like manner (as in teleosts). However, differences from teleosts were found that may have implications for the mechanisms for sound source location in azimuth. The common physiological characteristics among *A. fulvescens*, teleosts, and land vertebrates may reflect important functions of the auditory system that have been conserved throughout the evolution of vertebrates.

3:50

1pAB7. Sound source localization by fishes. Joseph A. Sisneros (Psych. Dept., Univ. of Washington, 337 Guthrie Hall, Seattle, WA 98195, sisneros@uw.edu)

The ability to locate sound sources enables animals to detect prey, avoid predators, and communicate with conspecifics, and is thus basic to the survival of many vertebrate species. Evidence suggests that the capacity to locate sound sources is common to amphibians, reptiles, birds, and mammals, but surprisingly it is not known whether fish locate sound sources in the same manner. Sound source localization by fishes continues to be an important topic in the hearing sciences, but the empirical and theoretical work on this topic has been contradictory and obscure for decades. Studies of sound localization by fishes have been difficult to conceive of primarily because fish are assumed not to use the same binaural acoustic cues as terrestrial animals, and secondarily, because the dominant theories for sound localization by fishes are rather complex and most fish are thought to be unable to detect the sound cues theoretically necessary for localization. Thus, the question of how fish locate sounds remains an open one. Dr. Sisneros' talk will describe the work of Fay and Sisneros that address sound source localization in the plainfin midshipman (*Porichthys notatus*), which has proven to be an exceptional species for fish studies of sound localization.

4:10

1pAB8. Not just fishes: Dick Fay's contributions to frog bioacoustics. Andrea M. Simmons (Brown Univ., Box 1821, Providence, RI 02912, Andrea_Simmons@brown.edu)

In addition to his many conceptual and technical contributions to fish bioacoustics, Dick Fay also guided and influenced research into the auditory worlds of anuran amphibians. In this talk, I will highlight the impact of Dick's scientific career on the understanding of frog bioacoustics. Topics covered will include the use of the shaker table system, which Dick designed to quantify directional hearing in fishes, to the study of pressure and particle motion sensitivity in bullfrog tadpoles; the importance of temporal coding in the auditory nervous system for mediating periodicity perception; and the use of psychophysical experiments for generating hypotheses about the evolution of hearing.

4:30

1pAB9. Comparison of psychoacoustic abilities in many vertebrate species. Glenis Long (Speech-Language-Hearing Sci., Graduate Ctr. of the City Univ. of New York, Graduate Ctr. CUNY, 365 Fifth Ave., New York, NY 10016, glong@gc.cuny.edu)

Psychoacoustic research with non-human species is very time consuming, but provides invaluable information about the underlying bases of hearing and its importance to humans and other species. Although most of Dick Fay's research has been evaluated the psychoacoustic abilities of one species of fish, he consistently compared this data with similar data from other vertebrates. He was not content to present data from one or two other species, but searched the literature to ensure that all possible comparisons were accessed to produce excellent figures combining this data. He made all the data he extracted from the literature available in an invaluable book (Fay, R. R. (1988). *Hearing in Vertebrates: A Psychophysics Databook*. Winnetka, IL: Hill-Fay Associates). This data are now being made available online on http://www.fayfoundation.org/category/hearing-in-vertebrates-a-psychophysics-databook/, a tool which will continue to stimulate research into the similarities and differences of psychoacoustic abilities in vertebrate species.

Contributed Paper

4:50

1pAB10. Efferent command may facilitate special frequency channel in the inner ear for vocal communication of Mongolian gerbils. Hiroshi Riquimaroux (Virginia Tech Int. Lab., Shandong Univ., 27 Shanda Nanlu, Jinan, Shandong 250100, China, hiroshi_riquimaroux@brown.edu)

Mongolian gerbil, *Meriones unguiculatus*, has been a good model animal for hearing, whose thresholds between 1 and 16 kHz are low. The animal also has a variety of repertoire in communication calls. Frequency range for 80% of communication calls was reported to be 30–50 kHz. However, the audiogram indicates that thresholds for 30–50 kHz are not low, not well suited for vocal communication. The present study examines if sensitivity for 30–50 kHz could be improved when necessary. We made a hypothesis that their attention might enhance inner ear sensitivity around 30–50 kHz for vocal communication. Then, frequency sensitivity of inner ear was examined by the cochlear microphonics (CM), comparing attended and unattended conditions. The findings suggest that attention to the companion animal may enhance CM power spectra around 30 kHz, especially for low intensities. Moreover, when a companion animal exists, even CM to short simple tone bursts of 25–40 kHz was enhanced. Findings from noise induced TTS experiments would also imply that CM responses around 30 kHz appeared to be amplified by the hair cell activities. [Work supported by Grant-in-Aid for Scientific Research (B) 26280064 from MEXT of Japan and Start-up grant from Shandong University.]

MONDAY AFTERNOON, 23 MAY 2016 SALON D, 5:30 P.M. TO 8:00 P.M.

Session 1pED

Education in Acoustics and Women in Acoustics: Listen Up and Get Involved

Tracianne B. Neilsen, Chair
Brigham Young University, N311 ESC, Provo, UT 84602

This workshop for Salt Lake City area Girl Scouts (ages 12–17) consists of a hands-on tutorial, interactive demonstrations, and a panel discussion about careers in acoustics. The primary goals of this workshop are to expose the girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please e-mail Traci Neilsen (tnb@byu.edu) if you have time to help with either guiding the girls to the event and helping them get started (5:00 p.m. to 6:00 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:00 p.m. to 7:30 p.m.). We will provide many demonstrations, but feel free to contact us if you would like to bring your own.

Session 1pID

Interdisciplinary and Student Council: Introductions to Technical Committees

Martin S. Lawless, Cochair
Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Anna C. Diedesch, Cochair
Hearing & Speech Sciences, Vanderbilt University, 7012 Sonya Drive, Nashville, TN 37209

Invited Papers

2:15

1pID1. An introduction to the acoustical oceanography technical committee. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca) and Christopher Bassett (Resource Assessment and Conservation Eng., National Marine Fisheries Service, Alaska Fisheries Sci. Ctr., Seattle, WA)

Commonly used air-side remote sensing techniques (e.g., cameras, GPS, wireless communications, and radar) are not suitable for many oceanographic applications because seawater is opaque to electromagnetic radiation. Acoustic signals, on the other hand, propagate relatively well underwater and can be used for remote sensing, imaging, and communications. The Acoustical Oceanography (AO) technical committee (TC) focuses on the development and use of acoustical techniques to understand geological, biological, chemical, and physical oceanographic parameters and processes. Research falling under the auspices of the AO technical committee is inherently interdisciplinary, overlapping with the Animal Bioacoustics, Signal Processing, and Underwater Acoustics technical committees. This presentation will introduce current AO research topics and the ways in which acoustical oceanography is advancing our knowledge of the ocean and its boundaries.

2:25

1pID2. Ever expanding applications in medicine and biology. Sanjay S. Yengul (Mech. Eng., Boston Univ., Ashland, MA) and Nathan McDannold (Radiology, Brigham and Women's Hospital, Harvard Univ., 221 Longwood Ave., Boston, MA 02115, njm@bwh.harvard.edu)

The Technical Committee on Biomedical Acoustics is one of the most diverse groups in the Acoustical Society of America. It is comprised of individuals whose interests cover a broad range of diagnostic and therapeutic applications or, more generally, the interaction of sound with biological materials. Exciting new developments in diagnostic ultrasound include innovative contrast agents and "dual-wave" imaging, examples being shear wave elastography, acousto-optic and opto-acoustic imaging. Therapeutic applications are expanding beyond lithotripsy and the treatment of tumors via thermal ablation. Mechanical ablation of soft tissue, known as histotripsy, is an area of active research. Transcranial magnetic resonance guided focused ultrasound is being used for the non-invasive treatment of various brain disorders, including essential tremor, neuropathic pain, and Parkinson's disease. The ability to temporarily open the blood-brain barrier and deliver chemotherapy to brain tumors using focused ultrasound has been demonstrated in a human for the first time. Researchers are also developing echogenic liposomes and microbubbles for targeted drug and gene delivery using low intensity focused ultrasound. As technology and protocols continue to evolve, developments in biomedical acoustics hold considerable promise towards the advancement of both diagnosis and treatment of debilitating diseases.

2:35

1pID3. The technical committee on musical acoustics: From log drums to synthesizers. Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

The Technical Committee on Musical Acoustics is home to researchers interested in the physics of musical instruments, music perception, and the science of the singing voice. Members of the committee typically come to musical acoustics from outside the field; therefore, there are a variety of backgrounds and interests represented. However, musical acousticians are united by a common interest in understanding the science of music. TCMU is an interdisciplinary technical committee with ties to architectural, psychological and physiological, signal processing, speech, physical, and structural acoustics.

2:45

1pID4. What is animal bioacoustics? Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

Animal Bioacoustics is the study of sound in animals. It covers: (1) animal sounds; communication, biosonar, and associated behavior; (2) sound production anatomy and neurophysiology; (3) auditory capacities and mechanisms, anatomy, and neurophysiology; (4) acoustic phylogeny, ontogeny, and cognition; (5) acoustic ecology, acoustic characterization of habitats, and effects of sound on animals; (6) passive acoustic tools and methods, hardware and software, for detection, classification, localization, tracking, density estimation, and behavior monitoring ; (7) active acoustic tools and methods, including animal-tracking sonars and echosounders, acoustic tags, pingers, deterrent devices, etc. Recent publications have treated various taxa: birds, terrestrial mammals (in particular, bats), marine mammals, amphibians, reptiles, fishes, insects, and crustaceans. Animal bioacoustics is beautifully interdisciplinary, and researchers have diverse backgrounds ranging from acoustics, via biology, engineering, mathematics, and physics, to zoology.

2:55

1pID5. Introduction to the Technical Committee on Physical Acoustics. Josh R. Gladden (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu)

The Technical Committee on Physical Acoustics (TCPA) of the Acoustical Society of America (ASA) includes scientists and engineers with an interest in the underlying physics of acoustical phenomenon. What unifies the members of TCPA is the approach to answering a particular question. The types of phenomena of interest to the physical acoustics community are fundamental acoustic wave propagation, including transmission, reflection, refraction, interference, diffraction, scattering, absorption, and dispersion of sound covering the entire frequency range from infrasound to ultrasound. There is also interest in the use of acoustics to study physical properties of matter, and to produce changes in these properties. TCPA currently has about 575 members with a primary interest in physical acoustics with 75–125 typically attending open TCPA meetings.

3:05

1pID6. The wide world of noise. Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Architectural Eng., Omaha, NE 68182-0816, eryherd@unl.edu)

Noise invades all aspects of our lives. The word noise is actually derived from the Latin word "nausea," with one possible connection being that unpleasant sounds were made by seasick passengers or sailors in ancient times. In modern times, the demand for noise research and consulting has intensified in concert with rising population densities, growing industrialized societies, escalating demands from consumers, and increasingly common standards and legislation related to noise. The Acoustical Society of America Technical Committee on Noise (TC Noise) is concerned with all aspects of noise, ranging from noise generation and propagation, to active and passive methods of controlling noise, to the effects of noise on humans and animals. This talk will explore the broad topic of noise and its impact on our world.

3:15

1pID7. An overview of research in underwater acoustics. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu)

Underwater acoustics (UW) studies underwater sound generation, propagation, and detection over scales ranging from millimeters to megameters, and often has overlapping interests with AB, AO, and SP. This talk will sample several topics of interest to the UW technical committee, including recent studies in propagation, inversion, and scattering, using both active and passive techniques. A long-term trend has been the discovery of how both deterministic and random components of the underwater acoustic field can be used to invert for surface, water column, sediment, and biological properties, which in turn provides parametrizations for improved "forward" modeling.

3:25

1pID8. Psychological and physiological acoustics: The encoding and interpreting of sound energy by biological systems. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

Hearing science, or the study of how the auditory system functions and responds to sound in humans and other animals, is a topic that has been studied using modern scientific methods for nearly 200 years. The current approaches to hearing science are highly multidisciplinary, integrating acoustics with many fields, such as psychology, biology, engineering, computer science, statistics, and audiology. The study of the encoding and perception of simple and complex sounds means that there is considerable overlap with other technical areas of the Acoustical Society of America, including architectural, speech, music, and animal bioacoustics. Hearing scientists are also concerned with hearing disorders, vibrotactile and vestibular sensation, the interaction of hearing with other sensory modalities, development, and aging, as well as learning and plasticity effects in auditory function. This talk will highlight (1) the various brain areas involved in auditory processing and (2) the technological innovations in hearing aids and cochlear implants that have made sound processing available to many people who, for most of human history, would have been either completely deaf or would have had only minimal awareness of sound.

3:35

1pID9. Signal processing: Ubiquitous in acoustics. James Preisig (JPAnalytics LLC, JPAnalytics LLC, 638 Brick Kiln Rd., Falmouth, MA 02540, jpreisig@jpanalytics.com)

The bat sends out a pulse of sound, hears the reflection off of an insect, and moves in for a meal. Similarly, a whale locates its prey with the reflections of sound. Both bats and whales also use reflected sounds to navigate and avoid obstacles. Humans hear speech and infer the message being send by the speaker. The listener to music gets the artistic message being sent by the composer and musician. Man made sounds in the ocean are used to communicate, locate objects, and monitor environmental conditions. The reflection of an ultrasound pulse off of a beating heart gives information to the doctor about the health of a fetus. These are but a few of the examples where "signal processing," by a machine, human and/or animal pulls information from a received acoustic signal. Signal processing is an important component of the activities and research conducted by the members of many technical committees in the ASA. The members of the Signal Processing in Acoustics Technical Committee, through theoretical development, implementation, and analysis study algorithms and acoustic signals in areas that span the society. This talk will present an overview of a few of these activities.

3:45

1pID10. Introduction to the speech communication technical committee. Benjamin Munson (Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, munso005@umn.edu)

The Speech Communication technical committee is one of the oldest and largest in the Acoustical Society of America. Our work covers three broad areas: speech production, speech perception, and computer speech processing. Studies presented at our meetings and in our journal cover languages spoken all over the world, including widely spoken languages like Mandarin and Spanish, and less widely spoken languages like Zoque and Tlingit. We study individuals across the lifespan, including both typical speakers and ones with speech, language, and hearing impairments. This brief presentation will highlight three recent presentations given at the society. These illustrate the breadth of our work, our centrality in the ASA, and our connections to the other technical committees.

3:55

1pID11. Introduction to the structural acoustics and vibration technical committee. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

The Structural Acoustics and Vibration Technical Committee (SAVTC) includes the study of motions and interactions of mechanical systems with their environments and the methods of their measurement, analysis, and control. This talk will provide a broad overview of the many research areas of interest to SAVTC. A few topics will be explored in more depth to provide background on some of the more common analysis methods used by members of the technical committee.

4:05

1pID12. Overview of the Engineering Acoustics Technical Committee. David A. Brown (ECE/ATMC/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

The Engineering Acoustics Technical Committee (EATC) and its members work to advance the methods, tools, and instrumentation used in the study of sound in all its disciplines. It is concerned with the development and improvement of measurement techniques, apparatus, new methods, acquisition of data, and the creation of knowledge in the field of acoustics. Some of the common areas involve: production and reception of sound, transducers and arrays, underwater and in-air acoustic systems, audio engineering and related electronics and amplification, instrumentation and recording, hearing aids, medical and industrial ultrasonics, noise monitoring and cancelation, materials and metamaterials. The talk also presents some hot topics in the EATC and some sample student research projects.

4:15

1pID13. Architectural acoustics—Hear here. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The discipline of Architectural Acoustics consistently produces more than 100 papers across six or more special sessions, at each meeting of the ASA. Student paper awards, student design competitions, and Knudsen lectures augment these activities. Joint sessions, particularly with noise, musical acoustics, psychological and physiological acoustics, and signal processing in acoustics, add more still to the architectural acoustics goings-on at every ASA conference. The sphere of influence is not limited to ASA alone, as TCAA members participate in the Green Construction Code of the International Code Council, Society of Motion Picture and Television Engineers Study Group: Movie Theater Sound System Measurement and Adjustment Techniques, Classroom Acoustics Standards, the American Institute of Architects Continuing Education System, and more. This busy committee also produces a steady stream of publications documenting recent work and deciphering standards for key stakeholders. Anyone with an interest in the field will find many opportunities to advance their own expertise, build a network of colleagues, friends and mentors, and contribute to the essential activities of the Technical Committee on Architectural Acoustics.

Contributed Paper

4:25

1pID14. How to submit and chair a special session. Michael V. Scanlon (RDRL-SES-P, Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

Organizing and chairing a special session at an Acoustical Society of America meeting is easy. The hardest part is raising your hand for the first time at a Technical Committee (TC) to promote a topic of interest to you and your peers. There are many reasons why proposing a focused Special Session can benefit the ASA membership and you. This presentation will describe the steps you need to take to propose a special topic, arrange for the session to be listed in the Call for Papers, and how to organize your session. You will understand how to contact specific subject matter experts to give invited presentations, and how to describe your topic to encourage other authors to contribute papers. This presentation will be especially helpful for students and first time meeting attendees, and will hopefully motivate more early-career members to attend TC meetings, volunteer to Chair upcoming special sessions, and increase interaction with subject matter experts in that particular area.

MONDAY AFTERNOON, 23 MAY 2016 SALON B/C, 1:30 P.M. TO 4:45 P.M.

Session 1pNS

Noise and ASA Committee on Standards: Community Noise II

Eric L. Reuter, Chair
Reuter Associates, LLC, 10 Vaughan Mall, Suite 201A, Portsmouth, NH 03801

Contributed Papers

1:30

1pNS1. Unexpected pros and cons of planned freeway sound wall removal. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

Sound walls are a common occurrence along heavily traveled freeways near sensitive receptor areas. As freeways lanes are increased to handle more traffic, vehicle speeds and traffic noise are prone to increase initially until traffic volumes increase to capacity. Under this scenario, impacted residents can petition for improved mitigation measures. A less common scenario occurs when sensitive receptor areas are abandoned and highway departments are petitioned to remove a freeway sound wall. This situation requires careful analysis and justification to be approved. If there are secondary sensitive receptors that have benefited from the reduced traffic noise due to a sound wall, what potential noise impacts might they experience if the freeway sound wall is removed? Recently, a commercial developer embarked on just such a project. Careful analysis of the proposed project and existing conditions revealed a more complex situation than originally described. Existing reflective sound walls of different heights on either side of the freeway, different near and far sensitive receptor areas behind the sound walls, different jurisdictions, topographical variations, and other competing commercial areas farther along both ends of the sound wall, all added to the project complexity and are discussed along with unintended consequences and unforeseen benefits.

1:45

1pNS2. Dynamic road traffic density estimation employing noise mapping with the use of grid supercomputing. Andrzej Czyzewski (Multimedia Systems Dept., Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland), Karolina Marciniuk, and Bozena Kostek (AudioAcoust. Lab., Gdansk Univ. of Technol., Narutowicza, 11/12, Gdansk, Pomorskie 80-233, Poland, karmarci@sound.eti.pg.gda.pl)

A noise prediction model of a large city agglomeration was elaborated in order to allow for a dynamic road traffic density estimation in vehicular networks. The implemented application adopts the model fed with traffic noise data based on frequently refreshed L_{DEN} levels. Calculations were made with the use of the numerical model developed for his purpose and then implemented on the PL-Grid supercomputing infrastructure. Data obtained through supercomputing and through the use of a standard noise map computing software were collated with measured levels acquired from the acoustic city monitoring system and then analyzed. The comparison performed afterwards shows a relatively good accuracy of the developed model. The numerical model of traffic noise and its main sources are briefly characterized. A full day dynamic noise map can be browsed as a set of 24 noise maps, one for each hour of the day which in turn allows for vehicular traffic density estimation based exclusively on acoustical data. [Research was subsidized by the Polish National Centre for Research and Development within the Grant No. OT4-4B/AGH-PG-WSTKT.]

2:00

1pNS3. Probability of receiving a blast noise complaint: Spatial analysis of complaint behavior around military installations. Dawn A. Morrison (US Army Corps of Engineers, ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61826), Jose Gonzalez (Statistics, Univ. of Illinois, Urbana-Champaign, Champaign, IL), and Edward T. Nykaza (US Army Corps of Engineers, ERDC-CERL, Champaign, IL, edward.t.nykaza@erdc.dren.mil)

This study explores the probability of receiving a blast noise complaint in response to military training noise. Community noise complaints are an ongoing impediment to military training and often result in training curfews and restrictions. As such, we want to understand where, by whom, and under what conditions complaints are used as a coping strategy in response to annoyance caused by military training noise. We build upon previous research conducted (Nykaza *et al.*, 2012), incorporating physical and human geography into the analysis to identify the situational context of the population subset that complains about military noise. Using both spatial and statistical analysis, we look at land use (LULC), census and socio-economic housing data paired with complaint and noise monitor data collected around one US Military Installation, to examine who (i.e., what subset of the population) is more likely to use complaint behavior as a coping strategy. Our results support the development of a noise complaint-forecasting model that military range managers can use to improve the coordination, communication, and relationships with surrounding communities.

2:15

1pNS4. Estimates of residential floor vibration induced by sonic booms. Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., MS 463, Hampton, VA 23681, j.klos@nasa.gov)

Future civil supersonic aircraft are expected to produce sonic boom noise within overflown communities that is substantially quieter than previous supersonic aircraft, which may allow regulators to replace the current prohibition on civil supersonic flight with a noise-based certification standard. In support of standards development, NASA has used an indoor simulation laboratory to investigate people's reaction to combinations of transmitted sonic boom signatures and related floor vibration levels. Given the limited extent of experimental data, numerical models were used to estimate the expected range in floor vibration level that people may experience within a large assortment of houses for various low-amplitude sonic booms from different aircraft. This presentation will discuss the predictive models, compare predicted floor vibration to measurements from field experiments in similar houses, and document the range of predicted floor vibration that informed a recent psycho-acoustic laboratory study. The analyses of subjective annoyance from this laboratory study are discussed in a companion presentation (Rathsam and Klos).

2:30

1pNS5. Vibration penalty estimates for indoor annoyance caused by sonic boom. Jonathan Rathsam and Jacob Klos (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jonathan.rathsam@nasa.gov)

Commercial supersonic flight is currently forbidden over land because sonic booms have historically caused unacceptable annoyance levels in overflown communities. NASA is providing data and expertise to noise regulators as they consider relaxing the ban for future quiet supersonic aircraft. One key objective is a predictive model for indoor annoyance based on factors such as noise and indoor vibration levels. The current study quantified the increment in indoor sonic boom annoyance when sonic booms can be felt directly through structural vibrations in addition to being heard. A shaker mounted below each chair in the sonic boom simulator emulated vibrations transmitting through the structure to that chair. The vibration amplitudes were determined from numeric models of a large range of residential structures excited by the same sonic boom waveforms used in the experiment. The analysis yielded vibration penalties, which are the increments in sound level needed to increase annoyance as much as the vibration does. For sonic booms at acoustic levels from 75 to 84 dB perceived level, vibration signals with lower amplitudes (+1 sigma) yielded penalties from 0 to 5 dB, and vibration signals with higher amplitudes (+3 sigma) yielded penalties from 6 to 10 dB.

2:45

1pNS6. On the use of a correction scale for the effect of intruding environmental sources at specified signal to noise ratios. Giora Rosenhouse (Acoust., SWANTECH Ltd., 9 Kidron St., Haifa 3446310, Israel, giora@swantech.co.il)

The British Standard BS 4142:2014 is a method for rating noise levels by comparing the rating level of the specific noise source with the existing background noise. This approach includes some corrections, but it does not properly consider the difference in frequency contents of both background noise and the examined specific noise. However, taking into account the influence of the spectrum of the intruding noise corrected by a scale that depends on the signal to noise ratio might solve the issue. It appears that the model of S/N influence on speech intelligibility and the influence of environmental background noise on the effect of specified noise source on people may have a common correction scale. If it is true, as measurements show, the human brain uses the same scale of S/N ratio for the perception of both speech and noise effects, as was shown by the author in the ASA meeting in Providence¸ 2014. This paper shows explicitly how the calculation is done in the case of noise compared with the calculation of the effect of S/N ratio on speech intelligibility. It includes a solved case of using of the correction scale.

3:00–3:15 Break

3:15

1pNS7. Results of a visitor survey of the acoustic environment at the Grand Canyon National Park. Pranav K. Pamidighantam and Paul D. Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, ppamidig@illinois.edu)

For the last decade, multiple studies have been conducted at national parks using the traditional uni-polar annoyance questions and the dose-response concept to determine the effects of aircraft noise on park visitors. This survey used pleasantness as a bipolar scale and breaks the hike into segments so as to understand the overall hike's assessment through its parts. This survey was conducted on three different trails at the Grand Canyon: the Hermit trail, a backcountry trail with a large amount of aircraft noise; the Widforss trail, a backcountry trail with minimal aircraft noise; and the Bright Angel trail, a maintained trail with a small amount of aircraft noise but many hikers. Helicopter noise is found to be the only major unpleasant noise. Segments for which no aircraft are noticed are judged to be primarily *very pleasant*. The "average" judgment with helicopters noticed falls between *neutral* and *slightly pleasant*, which is a drop of almost three categories out of a total of seven. The study finds that visitor perception of pleasantness of the acoustic environment at the Grand Canyon is significantly, negatively affected by helicopter overflights near trails.

3:30

1pNS8. Accounting for non-zero asymptotic annoyance prevalence rates in community tolerance level analyses. Vincent Mestre (LANDRUM & BROWN, 19700 Fairchild Rd., Ste. 230, Irvine, CA 92612, vmestre@landrum-brown.com), Richard Horonjeff (Acoust., Boxborough, MA), and Sandord Fidell (Fidell and Assoc., Woodland Hills, CA)

Dosage-response relationships between transportation noise exposure and the prevalence of noise-induced annoyance typically assume an asymptotic rate of annoyance of zero at low noise exposure levels. If the annoyance prevalence rate of any sigmoidal function at low exposure levels is non-zero, however, its predictions may not closely match field observations in low exposure ranges. Community tolerance level (CTL) analysis yields a dosage-response relationship that is easily modified to account for non-zero asymptotic annoyance prevalence rates. This presentation describes empirical analyses undertaken to investigate the effects on CTL values of field observations of annoyance at low levels of noise exposure.

3:45

1pNS9. Two weeks in the woods: Noise from a natural-gas compressor station. Thomas B. Gabrielson (Graduate Program in Acoust., Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

Natural gas development is permitted on Pennsylvania State Forest land; however, the Pennsylvania Bureau of Forestry is tasked not only with stewardship of natural resources but also with maintaining a natural environment for wildlife and recreational users of the land. As a precursor to human-impact studies of gas-compressor noise, a two-week long series of measurements was made near a natural-gas compressor station in the Tiadaghton State Forest in north-central Pennsylvania. Although several standard metrics were determined, the intent was to collect and analyze full time-series data at a number of different distances and directions from the station. Recordings were made in two bands: 5 to 10 000 Hz with 5 min of recording every hour for two weeks and 0.1 to 250 Hz recorded continuously for one week. Analyses of the narrowband spectral densities and of the spectrograms of the noise reveal specific source mechanisms and source variability. The high-fidelity recordings will be used subsequently both for listening tests with human subjects and exposure tests for wildlife. [Work supported by the Pennsylvania Department of Conservation and Natural Resources, Bureau of Forestry.]

4:00

1pNS10. Soundlapse: audio strategies for documenting soundscapes with timelapse photography. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

Timelapse photography—the procedure of taking photographs at regular intervals and then sequencing them together to produce a compressed portrayal of time passing—is widely popular across diverse areas such as scientific research, weather tracking, filmmaking, and even reality TV shows, which commonly use timelapse footage to segue between scenes and reflect the passing of time. The compelling visual effect and broad legibility of timelapse photography makes it a powerful tool for revealing flows, shifts, and patterns in the environment which would otherwise be too subtle to perceive in real time. One of the most important challenges in documenting the salient features of a soundscape is revealing similar shifts and patterns across time in the acoustic environment. However, temporal logic would suggest it is simply not possible to sequence such brief samples of the audio environment with any meaningful result. This project explores different audio capture and editing techniques to produce viable strategies for documenting the changes in a soundscape which accompany timelapse photography of the environment under study. Examples will be demonstrated, emphasizing the significant variables for adequately representing the elusive temporal characteristics of a soundscape.

4:15

1pNS11. Measurable domain for crowd noise in modern stadiums and arenas. Mojtaba Navvab (Univ. of Michigan, 2000 Bonisteel Blvd., Ann Arbor, MI 48109-2069, moji@umich.edu)

The noise generated by the organized fans or crowds during sporting events has created a challenge for sports facility management. The new demand for full compliance to national sports league rules on crowd noise, as well as health regulations ordinance on noise, require new methods in estimating and or measuring such environmental conditions. Given their dynamic range and possible classification, noise levels generated by large crowds have influenced the outcome of games. Recent analysis of the available data shows this situation has provided more excitement for the spectators and greater participation in the events. However, sound intensity of 95 to 110 dBA is now the typically reported range of recorded noise levels within various sports stadiums and arenas. This paper describes an integrated method utilizing ray tracing in simulating the sound path, and delay-sum beamforming in the time and frequency domains using the spherical wave concept, measuring the sound with 120 channel data recorders, and utilizing various acoustic software for data reduction, computer modeling, simulation, and analysis within ten selected sports facilities. The results show the associated room acoustic characteristics and the measurable domain for crowd noise and acoustic signature of modern stadiums and arenas that represent the significant component of all architectural elements with their contributions to the space including allowable noise dose as estimated for spectators' exposure to noise at a safe rate by NIOSH and NFPA 72 and compliance.

4:30

1pNS12. Noise dosimetry of ensemble musicians at Brigham Young University—Idaho. James Blaylock, Daxton Hawks, Ryan Flamm, William Weiss, and Jon P. Johnson (Phys., Brigham Young Univ. - Idaho, 525 S Ctr. St., Rexburg, ID 83460-0520, bla11009@byui.edu)

Students studying music can be exposed to high levels of sound while practicing, both the sound they create and the sound created by those around them. While playing one's instrument and creating sound are crucial to a student's success, the music itself and ambient sound may result in noise-induced hearing loss for the musician. We have made noise dose measurements to investigate the sound that students are exposed to while practicing in a large symphonic ensemble. Contributing factors of high volumes of sound may include students' instruments, a poorly insulated practice room, practice room resonances and reflection characteristics, and loud instruments being played in other rooms. Further research could include investigation of better insulated practice rooms, or possible rearrangements of musicians' seating.

MONDAY AFTERNOON, 23 MAY 2016 — SALON H, 1:30 P.M. TO 4:30 P.M.

Session 1pPA

Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Computational Methods in Physical Acoustics II

Amanda Hanford, Cochair
Pennsylvania State Univ., Applied Research Lab., PO Box 30 - MS 3230D, State College, PA 16804

D.Keith Wilson, Cochair
CRREL, U.S. Army ERDC, 72 Lyme Rd., Hanover, NH 03755-1290

Invited Papers

1:30

1pPA1. Numerical wave simulation for interactive audio-visual applications. Nikunj Raghuvanshi (Microsoft Res., 1 Microsoft Way, Redmond, WA 98052, nikunjr@gmail.com), Andrew Allen (Qualcomm Inst., Univ. of California San Diego, San Diego, CA), and John Snyder (Microsoft Res., Redmond, WA)

Sound is an integral part of many interactive applications, such as 3D computer games. The generality and complexity of the simulated domains makes wave simulation a natural fit. The opposing goals of real-time execution and compute intensiveness of wave simulation provide a rich ground for designing novel computational methods. Two instances of such audio-visual interactive techniques will be discussed. The first technique employs precomputed wave simulation for providing acoustic effects in computer games. Time-domain 3D simulation is performed from many prospective player positions, resulting in a 7D field containing billions of acoustic responses. Perceptual encoding is then performed to drastically reduce memory requirements. Interactive motion of player and sources in the game can then be accommodated via quickly decoding and auralizing the pre-calculated wave effects. The second technique explores the limits of real-time parallel numerical acoustic computation on today's desktop graphics cards. The user is presented with a 2D "sandbox" environment for designing virtual 2D musical wind instruments that are instantly simulated in real-time for the entire human auditory bandwidth. Instrument shape editing and performance via manipulating tone-holes/valves can be done on-the-fly. Video demonstrations of both techniques will be shown along with a discussion of the core technical ideas.

1:50

1pPA2. Ellipsoidal perfectly matched layers for acoustics. Greg Bunting, Arun Prakash (School of Civil Eng., Purdue Univ., West Lafayette, IN), Clark Dohrmann, and Timothy F. Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., PO Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov)

In this talk, we present a novel ellipsoidal formulation and massively parallel implementation of a perfectly matched layer (PML) for acoustics and structural acoustics. Perfectly matched layers and infinite elements are commonly used for finite element simulations of acoustic waves on unbounded domains. PML is a more recent technology that is gaining popularity due to ease of implementation and effectiveness as an absorbing boundary condition. In this study, we extend well-known curvilinear formulations for PML to ellipsoidal domains, thus allowing for a minimal volume encapsulation of the structure of interest. We discuss the issues involved in parallelization and compare the performance of PML against infinite elements on a set of representative acoustic problems on exterior domains. We examine the conditioning of the linear systems generated by the two techniques by examining the number of Krylov-iterations needed for convergence to a fixed solver tolerance. We also examine the scalability of the methods in terms of the number of Helmholtz solver iterations as the number of PML layers and infinite element order are increased. [Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL850000.]

2:10

1pPA3. Time-domain propagation in porous media using the auxiliary differential equation method: Application for outdoor acoustics. Didier Dragna, Pierre Pineau, and Philippe Blanc-Benon (Ctr. Acoustique, Laboratoire de Mécanique des Fluides et d'Acoustique, LMFA UMR CNRS 5509 - Ecole Centrale de Lyon, 36, Ave. Guy de Collongue, Ecully 69134, France, didier.dragna@ec-lyon.fr)

Time-domain sound propagation equations in porous media usually involve convolutions, which can be computationally cumbersome. Indeed, a naïve approach would require to store the acoustic variables at every time-step, which would lead to an unacceptable memory space for long range propagation. To reduce the computational burden associated to convolutions, an efficient numerical method, referred to as the auxiliary differential equation (ADE) method, originated from the computational electromagnetic community and presented in Dragna *et al.* [J. Acoust. Soc. Am. **138**, 1030–1042 (2015)] is proposed. The idea behind is to approximate the relaxation functions, which depend on the porous media properties, by rational functions in the frequency-domain. The time variation of the convolution is thus given by first-order differential equations which can be solved using standard time-marching schemes. The accuracy of the method is investigated and compared to that of recursive convolution methods. The ADE method is then applied for outdoor sound propagation using the time-domain rigid-frame model proposed by Wilson *et al.* [Appl. Acoust. **68**, 173–200 (2007)] in the ground. Results obtained with Wilson's model are compared at long range to those obtained with Zwikker and Kosten's equations and with an equivalent impedance surface for different flow resistivities.

2:30

1pPA4. The Iterative Nonlinear Contrast Source method: A versatile approach for numerical computation of nonlinear acoustic wave fields. Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628CJ, Netherlands, m.d.verweij@tudelft.nl)

Acoustic wave propagation is inherently nonlinear because the acoustic field influences the state of the medium. When frequency, amplitude, and propagation distance are sufficiently high, nonlinear effects become significant. Although undesired in many cases, nonlinearity is explicitly employed in medical ultrasound, e.g., by using reflections of the emerging higher harmonics to improve diagnostic image quality. Simulation of novel ultrasound devices and protocols requires the computation of pulsed nonlinear ultrasound fields over three dimensional domains that are typically hundreds of wavelengths in size. The Iterative Nonlinear Contrast Source (INCS) method primarily has been developed for this purpose. It solves the Westervelt equation by considering its nonlinear term as a contrast source that radiates, alongside the primary sources, into a linearized homogeneous background medium. Convolution with the analytical background Green's function yields the acoustic pressure field of the sources. An increasingly accurate approximation of the nonlinear field is obtained by iteratively substituting the computed field into the nonlinear contrast source. Spatiotemporal filtering allows the four-dimensional convolutions to be performed by FFTs that employ only two points per wavelength. The talk will cover the basic features of INCS, inclusion of attenuation and medium inhomogeneity, presentation of some characteristic results, and possible future extensions.

2:50–3:10 Break

3:10

1pPA5. Rapid computation of waves diffracted by an absorbing plane with precise error bounds. Kai M. Li and Yiming Wang (Mech. Eng., Purdue Univ., 140 South Russell St., West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

The current study is motivated by a recent paper [L. N. Trefethen and J. A. C. Weideman, "The exponentially convergent trapezoidal rule," SIAM Rev., **36**, 385–458 (2014)]. It is well known that waves diffracted by an absorbing plane due to a point or line source can be expressed in Fourier integrals. At low frequencies and short ranges, numerical solutions may be obtained by evaluating the integral, for example, by the fast field formulation (FFP). However, it is found prohibitive expensive to use FFP at high frequencies and long ranges. On the other hand, the solutions can be efficiently evaluated along the steepest descent path at all situations. A common approach involves expansion of the kernel function about the saddle point and application of the pole subtraction method. By retaining the first term of the Maclaurin series, this method gives a fairly accurate analytic solution. This paper explores the trapezoidal rule for evaluating the Fourier integral along the steepest descent path with a prescribed precision. An error bound analysis has been conducted to confirm the predicted accuracy. The required computational time is comparable to those calculated by the asymptotic solution except that the trapezoidal rule yields even more accurate numerical results.

Contributed Papers

3:30

1pPA6. Linking the viscous grain-shearing mechanism of wave propagation in marine sediments to fractional calculus. Vikash Pandey and Sverre Holm (Dept. of Informatics, Univ. of Oslo, Postboks 1080, Blindern, Oslo 0316, Norway, vikashp@ifi.uio.no)

This work is an extension of Pandey and Holm [Abstract, JASA (2015)] and builds on Buckingham's viscous grain-shearing (VGS) model [JASA (2007)]. The VGS model is an extension to the grain-shearing mechanism initially proposed to explain wave propagation in marine sediments. We find that the material impulse response function obtained from the VGS model is characterized by power-law terms which are inherent to the framework of fractional calculus. The VGS model in the fractional framework yields two equations; a fractional-order wave equation for the compressional wave and a modified fractional-order diffusion-wave equation for the shear wave. The order of the fractional equations as well as the two characteristic time constants are identified as the physical parameters of the medium. The viscous dissipation incurred due to inter-granular sliding across the pore-fluid modifies the low frequency behavior of the dispersion plots significantly. The low frequency behavior (<1 Hz) for both compressional and shear wave show normal dispersion of decreasing phase velocity with increasing frequency. These features were not observed in the original VGS model. The overall goal is to establish a physically based fractional framework, which can effectively incorporate the role of different material parameters in wave propagation.

3:45

1pPA7. Three-dimensional vibrations of acoustoelectric superlattice in ferroelectric plate. Andriy Nadtochiy (Taras Shevchenko Univ., Kyiv, Ukraine), Lucien Cremaldi (Phys. and Astronomy, Univ. of MS, Oxford, MS), and Igor Ostrovskii (Phys. and Astronomy, Univ. of MS, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu)

The vibrations and matching acousto-electric resonances in 3-D periodically poled $LiNbO_3$ bar are investigated. Two superlattices are fabricated in the z-cut 0.5-mm-thick wafers. They consist of the inversely poled domains of 0.45- or 0.3-mm long along the x-axis and tens-mm wide along the y-axis. Metal electrodes are deposited at the ends of superlattices along the x-axis. Experimentally, the vibrations are excited in MHz frequency range by the electric current along the x-axis or acoustic burst propagating along this axis. The function generator, digital oscilloscope, and spectrum analyzer allow measuring frequency dependencies of the acousto-electric resonances in rf-admittance Y(f), and transmission coefficient for acoustic burst propagating through the superlattice. The finite element method computations yield the acoustic displacements and acousto-electric admittance versus frequency. The displacements have three components along the x, y, and z axes. They are minimized within the stop-bands frequencies, which may vary for different displacement components. The number of resonances in Y(f) occur from 2 to 8 MHz. The spectral positions of resonances correlate with the frequencies of stop-bands boundaries for all three displacement components. The results obtained may be used for development of new applications such as multi-displacement ultrasonic transducers and actuators, acousto-electric filters, and oscillators.

4:00

1pPA8. Towards real-time two-dimensional wave propagation for articulatory speech synthesis. Victor Zappi, Arvind Vasudevan, and Sidney Fels (Elec. and Comput. Eng., Univ. of Br. Columbia, 2-2300 E 11th, Vancouver, BC V5N 1Z8, Canada, victor.zappi@gmail.com)

The precise simulation of voice production is a challenging task, especially when real-time performances are sought. To fulfill real-time constraints, most articulatory vocal synthesizers have to rely on highly simplified acoustic and anatomical models, based on 1D wave propagation and on the usage of vocal tract area functions. In this work, we present a 2D propagation model, designed to simulate the air flow traveling through the midsagittal contour of the vocal tract. Building on the work by Allen *et al.* [Andrew Allen and Nikunj Raghuvanshi, "Aerophones in flatland: Interactive wave simulation of wind instruments," ACM Trans. Graph. **34**, Article 134 (2015)], we leverage OpenGL and GPU parallelism for a real-time precise 2D airwave simulation. The domain is divided into cells according to a Finite-Difference Time-Domain scheme and coupled with a self-oscillating two-mass vocal fold model. To investigate the system's ability to simulate the physiology of the vocal tract and its aerodynamics, two studies are presented. First, we compare the performances in vowel production of our 2D approach with other 1D wave propagation systems in literature, using area functions. Subsequently, this case is extended by replacing area functions with 2D vocal tract contours derived from 3D MRI data.

4:15

1pPA9. An acoustics library for the Modelica multidomain modeling language. Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16803, sct12@psu.edu)

The Modelica® modeling language is especially well-suited for modeling the complicated electrical, mechanical, magnetic, and acoustical connections within an acoustic transducer or other acoustic system. The Modelica Standard Library includes many of the components that are needed for modeling electromechanical systems, but does not include piezoelectric or acoustical components. This paper describes methods and modeling code to provide an acoustical domain and several methods to include piezoelectric length expander pieces in the Modelica language.

Session 1pSAa

Structural Acoustics and Vibration and Musical Acoustics: Analysis of Vibration Based Musical Instruments

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Brian E. Anderson, Cochair
N145 Esc, Brigham Young Univ., MS D446, Provo, UT 84602

Invited Papers

1:00

1pSAa1. Detection of longitudinal resonances in brass wind instrument bells. Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu) and Wilfried Kausel (Inst. of Music Acoust., Univ. of Music and Performing Arts, Vienna, Austria)

It was shown more than a decade ago that the vibrations of the bells of brass wind instruments can affect the sound produced by the instrument. The effect of damping the vibrations is easily perceived by both the player and the listener, but the change in the sound cannot be attributed to merely eliminating direct radiation from the structure. Recently, it has been shown that coupling between the air column and the longitudinal motion of the bell is partially responsible for this effect, and it is possibly the dominant cause. However, identifying the longitudinal resonances of a flaring circular duct and separating them from the elliptical resonances is challenging. Using a combination of laser Doppler vibrometry and common-mode rejection electronic speckle pattern interferometry allows these modes to be unambiguously identified.

1:20

1pSAa2. The musical saw and the flexatone: An experimental study of confined vibrational modes in metal plates of variable curvature. Randy Worland (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

The musical saw and the flexatone are two quite different musical instruments that rely on similar physical principles for their sound production. The musical saw typically consists of an ordinary sawblade bent by the player into an "*S*"-shape and then bowed, producing a very pure tone. Skilled players produce melodies, vibrato, and glissando effects by dynamically varying the blade's curvature as well as the bowing location. The flexatone is a percussion instrument in which wooden beaters strike a flexible metal sheet. The resulting sounds can again be modulated by varying the curvature of the metal with slight hand pressure. Both instruments rely on the confinement of vibrational bending modes between regions of blade curvature. These confined vibrational patterns do not extend to the ends of the blade, and are therefore not influenced by the end conditions, including the damping that would otherwise be caused by the player's hand. The mode shapes and frequencies of these resonances are investigated as a function of blade geometry and curvature, using electronic speckle-pattern interferometry.

1:40

1pSAa3. Reconstructing the piano hammer force from measurements and filtering of the string velocity. Antoine Chaigne (Institut für Wiener Klangstil (Music Acoustics), Univ. of Music and Performing Arts (MDW), Anton-von-Webern Platz 1, Vienna 1030, Austria, antchaigne@gmail.com)

A method is presented for reconstructing the piano hammer force from measurements and filtering of the string velocity. The construction of the filter is based on the propagation of the pulses generated by the hammer blow in two opposite directions along the string. For the first five octaves of most pianos, and for a measuring point situated between hammer and agraffe, the hammer force can be derived by considering the incoming wave from the hammer and its first reflection by the agraffe only. For the higher notes, 4- or 8-waves schemes must be considered in order to reconstruct the hammer force over the complete range of the piano. The theory is first validated on simulated string velocities, by comparing imposed and reconstructed hammer forces. The simulations are based on a nonlinear damped stiff string model (JASA **134**, pp. 648–665). In a second step, the method is applied to string velocities measured of real pianos. Its efficiency is further validated through simulation of the string velocity with the reconstructed hammer force at the input, and comparison with the measured velocity. [Work supported by the Lise-Meitner-Fellowship M1653-N30 of the Austrian Science Fund (FWF).]

2:00

1pSAa4. Measuring the low-frequency response of an acoustic guitar. Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

The low-frequency response of an acoustic guitar is strongly influenced by the combined behavior of the air cavity and the top plate. The sound hole–air cavity resonance (often referred to as the Helmholtz resonance) interacts with the first elastic mode of the top plate creating a coupled oscillator with two resonance frequencies that are shifted away from the frequencies of the two original, uncoupled oscillators. This effect has been modeled using finite elements for the top plate and boundary elements for the air cavity with rigid sides and back and no strings. The natural frequencies of both the individual and combined oscillators will be compared to measurements. The modes and natural frequencies of the coupled and uncoupled top plate were obtained using experimental modal analysis. An array of electret microphones was used above the sound hole to compare the phase of acoustic pressure with that of the top plate vibration. The uncoupled acoustic mode was then measured after placing sandbags on the top plate to restrict its motion. The measurements confirm the mode shapes, natural frequencies, and damping of the model.

MONDAY AFTERNOON, 23 MAY 2016 — SALON J, 2:40 P.M. TO 4:55 P.M.

Session 1pSAb

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Contributed Papers

2:40

1pSAb1. Experimental active control of cylindrical shells using the weighted sum of spatial gradients control metric. Pegah Aslani, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N203 ESC Brigham Young University, Provo, UT 84602-4673, pegah.aslani@gmail.com), and Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

Cylindrical shells are often used in industry and there are many applications in which it is desired to reduce the sound radiated from the vibrating structure. At low frequencies, active structural acoustic control (ASAC) is able to offer solutions for this purpose. The technique of using the weighted sum of spatial gradients (WSSG) as a control metric has been developed previously for cylindrical shells as well as for flat structures. This metric has been demonstrated to be correlated with the radiated sound power and is therefore effective in attenuating the radiated sound. The method has also been shown to be robust with respect to the error sensor location. This paper will discuss the experimental implementation of WSSG on cylindrical shells and show experimental control results that have been obtained. The effectiveness of the method has been investigated by comparing the attenuation of the radiated sound power that is achieved both in the numerical model and the experimental results. The radiated sound power in the model is calculated through using the radiation modes for cylindrical shells, while for the experiment the sound power is measured using the ISO 3741 standard. A comparison of the experimental and numerical results will also be presented. [Work supported by NSF.]

2:55

1pSAb2. Active control of fluid-loaded vibrations in the mammalian inner ear. Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, julien.meaud@me.gatech.edu)

The mammalian inner ear is a sensory system that converts acoustic signals into electrical signals sent by hair cells to the brain. Hair cells are stimulated by vibrations of cochlear structures that are immersed in cochlear fluids. Normal hearing requires the presence of a nonlinear active feedback mechanism that boosts vibrations of cochlear structures and makes it possible to achieve high sensitivity, high frequency selectivity, and a broad dynamic range. In this talk, I will present the key characteristics of active control of vibration in the mammalian inner ear. I will demonstrate that computational models that take into account the structural acoustics of the inner ear and include realistic models of the active feedback mechanism can help to better understand the active control of vibrations in the inner ear. This understanding could be exploited to design biomimetic sensors and acoustic filters.

3:10

1pSAb3. Development of smart noise control technology. Lingguang Chen and Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, lingguang.chen@wayne.edu)

Effective noise control requires correctly identifying the components of structural vibrations that can actually produce sound. Standard practices in engineering applications include experimental modal analysis, transfer path analysis, etc., that are ad hoc in nature and oftentimes ineffective at all. This paper presents a new strategy that aims at identifying and suppressing the most critical forced-vibro-acoustic components (F-VACs) of vibrating structures that are directly responsible for structure-borne sound radiation. The F-VACs may be extracted from the transfer function that correlates the normal surface velocity of a vibrating structure to the resultant sound pressure, and their relative contributions be identified by singular value decomposition. Unlike transfer path analysis, the transfer function in this case can be obtained theoretically through the HELS-based NAH, rather than experimentally, thus significantly improving the effectiveness of this approach. Moreover, a laser vibrometer is used to collect input data, which greatly simplifies the complexity involved in data acquisition process of traditional NAH. Experimental validations of using this new strategy to reduce noise emission from an arbitrarily shaped vibrating structure are presented, and its effectiveness is compared with that of experimental modal analysis.

3:25

1pSAb4. The sound of an elastic airfoil in the wake of a vortex generator. Avshalom Manela (Aerosp. Eng., Technion, Technion City, Haifa 32000, Israel, amanela@technion.ac.il)

The acoustic signature of an acoustically compact tandem airfoil setup in high-Reynolds number flow is investigated. The upstream airfoil is rigid and is actuated at its leading edge with small-amplitude sinusoidal pitching motion. The downstream airfoil is taken passive and elastic, with its motion forced by the vortex-street excitation of the upstream airfoil. The near-field description is obtained via potential thin-airfoil theory. It is then applied as a source term into the Powell-Howe acoustic analogy, to yield the far-field dipole radiation of the system. To assess the effect of downstream-airfoil elasticity, results are compared with calculations for a non-elastic setup, where the downstream airfoil is rigid and fixed. Depending on separation distance between airfoils, airfoil-motion and airfoil-wake dynamics shift between in-phase (synchronized) and counter-phase behaviors. Downstream airfoil elasticity acts to either amplify or suppress sound, through the direct contribution of elastic-airfoil motion to the total signal. Resonance-type motion of the elastic airfoil is found when the upstream airfoil is actuated at the least stable eigenfrequency of the downstream structure. This, again, results in system sound amplification or suppression, depending on separation distance between airfoils. With increasing actuation frequency, the acoustic signal becomes dominated by the direct contribution of upstream airfoil motion, whereas the relative contribution of the elastic airfoil to the total signature turns negligible.

3:40

1pSAb5. Driver placement for selective modal excitation in flat-panel loudspeakers. Michael C. Heilemann, David Anderson, and Mark F. Bocko (Elec. and Comput. Eng., Univ. of Rochester, 500 Joseph C Wilson Blvd., 405 Comput. Studies Bldg., Rochester, NY 14627, mheilema@ur.rochester.edu)

The audio quality of flat panel loudspeakers may be improved dramatically by selectively driving specific panel modes using an array of force actuators distributed on the panel surface. An optimal actuator array layout will maximize the energy coupling between the actuators and selected panel modes, while simultaneously eliminating modal crosstalk within the operating frequency band. We present results for optimal force actuator array layouts for both dynamic force actuators and piezoelectric patch actuators. Experiments on panel loudspeakers using optimized force actuator array layouts show that each of the panel modes within the tuning bandwidth may be independently addressed, and that the optimized array can limit the intermodal crosstalk between -17 dB and -22 dB within the bandwidth using non-ideal force transducers. The methods described may also be useful in structural vibration control and noise reduction.

3:55

1pSAb6. Measures of vibrational localization on point-driven flat-panel loudspeakers. David Anderson, Mark F. Bocko, and Michael Heilemann (Elec. Eng., Univ. of Rochester, 526 Comput. Studies Bldg., Rochester, NY 14627, dander22@ur.rochester.edu)

Previous publications on flat-panel, or distributed-mode, loudspeakers generally assume that a localized driving force is able to spread energy evenly across the surface of a panel. However, investigations have shown that panel vibrations remain localized around the driving point at high frequencies, and this paper presents a deeper investigation into this phenomenon. Energy spreading will only occur when the panel is actuated in a frequency region with a low density of modes, as many modes actuated together will combine to form a band-limited delta function at the location of the driving force. A quantitative measure of localization is introduced, based on the ratio of the energy contained in a small region around the driving point to the energy contained in the entire panel. Simulations demonstrate that the frequency cutoff for localized vibrational behavior is dependent on the damping rate of the modes and the location of the driver. Experiments validate this theory by analyzing the vibrational behavior of a plate subject to small inertial drivers at various locations with a laser vibrometer.

4:10

1pSAb7. Nonlinear damage detection and localization via an innovative metamaterial-based sensor. Marco Miniaci (Laboratoire Ondes et Milieux Complexes, UMR CNRS 6294, Univ. of Le Havre, 22 Rue Bellot, Le Havre 76600, France, marco.miniaci@gmail.com), Anastasiia Krushynska, Federico Bosia (Dept. of Phys., Univ. of Torino, Torino, Italy), Antonio Gliozzi, Marco Scalerandi (Dept. of Appl. Sci. and Technol., Politecnico di Torino, Torino, Italy), Bruno Morvan (Laboratoire Ondes et Milieux Complexes, UMR CNRS 6294, Univ. of Le Havre, Le Havre, France), and Nicola Pugno (Dept. of Civil, Environ. and Mech. Eng., Univ. of Trento, Trento, Italy)

In recent years, acoustic metamaterials have attracted increasing scientific interest for very diverse technological applications ranging from sound abatement to ultrasonic imaging, mainly due to their ability to act as band-stop filters. At the same time, the concept of chaotic cavities has been recently proposed as an efficient tool to enhance the quality of nonlinear signal analysis, particularly in the ultrasonic/acoustic case. The goal of the present work is to merge the two concepts to propose a metamaterial-based device that can be used as a natural and selective linear filter for the detection of signals resulting from the propagation of elastic waves in nonlinear solids, e.g., in the presence of damage, and as a detector for the damage itself in time reversal experiments. Numerical simulations and experimental measurements based on scanning laser Doppler vibrometer demonstrate the feasibility of the approach and the potential of the device in providing improved signal to noise ratios and enhanced focusing on scatterer locations.

4:25

1pSAb8. Wave propagation through additive manufactured phononic crystals. Peter Kerrian (Graduate Program in Acoust., The Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, pak215@psu.edu), Amanda Hanford, Corey Dickman, and Dean Capone (Appl. Res. Lab Penn State Univ., University Park, PA)

Obtaining the effective parameters is an important step in designing a unit cell for a variety of metamaterial applications. The effective parameters including the density and bulk modulus govern how a wave propagates through a medium constructed from unit cells. Additive manufacturing has created the potential to change the static density of materials by encapsulating metallic powder inside the produced component. This work will use modal analysis to extract the bulk modulus from additive manufactured rods. After the material properties are determined, numerical and experimental results will be compared for wave propagation through a phononic crystal constructed from additive manufactured rods of different densities.

4:40

1pSAb9. Calculation of flanking sound transmission according to ISO 15712-1—Comparison between the simplified and the detailed method. Christoph Hoeller, Jeffrey Mahn (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca), and David Quirt (JDQ Acoust., Ottawa, ON, Canada)

The 2015 edition of the National Building Code of Canada (NBCC) specifies sound insulation requirements between dwelling units in terms of Apparent Sound Transmission Class (ASTC). The ASTC includes both the direct sound transmission through the separating element between adjacent rooms as well as the sound transmission via flanking paths. One of the ways to establish compliance with the NBCC involves a calculation procedure based on ISO 15712-1, in which the flanking sound transmission is predicted from the measured sound transmission through individual building elements combined with the attenuation at their junction. The calculation can be performed in third-octave bands ("Detailed Method") or using single-number ratings such as the STC ("Simplified Method"). This presentation will describe the two calculation procedures, before focusing on the differences between. In extended studies at the National Research Council Canada, it was found that the simplified method sometimes leads to misleading results. An alternative method for calculating the ASTC of walls with linings was proposed, which ensures that the simplified method yields more conservative results than the detailed method. To achieve the best possible estimate of the sound insulation performance of buildings systems with linings, the detailed method should be used.

Session 1pSC

Speech Communication: Acoustics and Perception of Speech (Poster Session)

Tuuli Morrill, Chair
George Mason University, 4400 University Dr., Fairfax, VA 22030

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

Contributed Papers

1pSC1. Experience-dependent plasticity in the neural weighting of pitch dimensions: A machine learning approach. Bharath Chandrasekaran, Rachel Reetzke, Han-Gyol Yi, Jessica Roeder, Zilong Xie, and W.Todd Maddox (Univ. of Texas at Austin, 1 University Station, Austin, TX 78712, bchandra@austin.utexas.edu)

We conducted a cross-linguistic study to evaluate the impact of language experience on midbrain encoding of acoustic dimensions. Midbrain electrophysiological responses were recorded to the four Mandarin tones in native Chinese (N = 10) and English (N = 10) listeners, through a counter-balanced block design. English participants were trained over multiple days to achieve tone categorization accuracy and reaction time equal to that of the Chinese participants. We assessed the extent to which the four Mandarin tones could be discerned from the electrophysiological responses, using a data-driven machine learning approach. The machine learning output was used to generate dissimilarity matrices that were subjected to a multidimensional scaling (MDS) model. A two dimensional MDS solution emerged that corresponded to "pitch height" and "pitch direction" of the Mandarin tones. Findings derived from the individual differences scaling (INDSCAL) method revealed that, initially, pitch direction was weighted more by the Chinese participants relative to the English participants. However, following training, relative weighting on pitch direction was found to increase in the English participants, comparable to the Chinese participants. These results suggest that long-term language experience and short-term training with Mandarin tones selectively enhances weighting of pitch dimensions at the level of the auditory midbrain.

1pSC2. Is perceptual learning influenced by inherent variability of phonetic category? Reiko Kataoka and Hahn Koo (Linguist and Lang. Development, San José State Univ., One Washington Square, San Jose, CA 95192-0093, reiko.kataoka@sjsu.edu)

A body of research has shown the effect of lexically guided perceptual learning (e.g., Norris *et al*., 2003), whereby listeners modify phonetic category boundaries after being exposed to critical words that contain segments that are phonetically ambiguous and serve as new exemplars of a target phonetic category. It has been shown that several factors constrain the extent of learning, and one recent study (Stevens *et al*., 2007) suggests that inherently variable phonetic categories might resist retuning. To test this idea, we examined modifiability of two vowel categories, [i] and [u], the latter of which has a wider range of variability than the former due to contextual variation as well as ongoing sound change. Our listeners heard exposure stimuli that were ambiguous between [i] and [u], embedded in lexical frames that yield real words only if the vowel is /i/ (/i/-words) or /u/ (/u/-words), during lexical decision tasks. Each listener's category boundary between [i]-[u] was measured Pre- and Post-exposure. Our results show that listeners tend to shift boundaries more for [i] than [u], suggesting a way that listeners balance between flexibility and stability in speech perception.

1pSC3. Phonemic category formation and suppressed sensitivity to extraneous acoustic cues. John Matthews (Faculty of Letters, Chuo Univ., 742-1 Higashi Nakano, Hachioji, Tokyo 192-0393, Japan, matthews@tamacc.chuo-u.ac.jp), Takako Kawasaki, and Kuniyoshi Tanaka (Faculty of Letters, Hosei Univ., Ichigaya, Tokyo, Japan)

L2 learners are known to be sensitive to acoustic cues beyond the bare minimum required to distinguish target language phonemic contrasts. This paper examines changes in this sensitivity to such extraneous details in the acoustic signal following extended exposure to L2 input. Our Category Activation Threshold Model (CATM) posits that the acquisition of new phonological categories brings about inhibitory mechanisms that actively suppress the processing of acoustic cues not needed for phonemic distinction. To assess the use of acoustic detail at different levels of proficiency, we administered a similarity judgment task to two groups of Japanese learners of English: university students with no study-abroad experience (NoSA: n = 22), and students with 3–12 months experience living and studying in an English-speaking country (SA: n = 18). SA participants judged the acoustic distance between pairs with one native and one non-native segment, /s/~/θ/, /z/~/ð/, to be greater than NoSA participants did. However, the SA group rated the acoustic difference between two utterances of the same stimulus item containing non-native /f/ or /v/ to be smaller than the No-SA group did. This result accords with the CATM, exhibiting the inhibitory effects of category formation.

1pSC4. Relationship between perceptual accuracy and information measures: A cross-linguistic study. Shinae Kang and Keith Johnson (Linguist, UC Berkeley, 1203 Dwinelle Hall, UC Berkeley, Berkeley, CA 94720-2650, sakang2@berkeley.edu)

Variation in the amount of information transmitted by speech elements correlates with (and helps explain) a number of speech patterns, such as phonetic reduction during production and sound change. However, the cross-linguistic effect of information content on speech perception is relatively understudied. This study fills this gap and investigates the relationship between perceptual accuracy and two information measures in English, Korean, and Japanese. We computed the informativity (weighted negative contextual predictability) and functional load (the contribution of a speech element to lexical contrast) of /p/, /t/, and /k/ in sub-syllabic speech chunks in English, Korean, and Japanese sub-lexical corpora. Native speakers of the three languages identified consonant clusters in VC.CV stimuli in a perception experiment. A logistic model predicting perceptual accuracy from informativity and functional load found that informativity is generally negatively correlated with perceptual accuracy, while functional load is positively correlated. Furthermore, the strength of this correlation is different for the three languages studied. We interpret the result as reflecting differences in syllable structure across the three languages.

1pSC5. Perceptual compensation in nasal and nasalized vowels in Brazilian Portuguese. Luciana Marques (Linguist, Univ. of Colorado, Hellems 290 295 UCB, Boulder, CO 80309, luciana.marques@colorado.edu)

Listeners of a language can factor out effects of coarticulation in the signal, consistent with the language's phonological patterns. Such perceptual compensation should not occur for phonetic features that are inherent to a segment, not effects of coarticulation. Vowel nasality is an interesting feature in this case, because it can be coarticulatory or inherent depending on the language. In Brazilian Portuguese (BP), it can be both. Contrastive nasal vowels may be followed by a brief nasal resonance (nasal appendix), while coarticulatory nasalized vowels must be followed by a nasal consonant. This study explores whether or not BP listeners perceptually compensate for vowel nasality in the perception of both phonemic nasal and coarticulatory nasalized vowels in BP, examining. If BP listeners process nasal vowels as phonemes, they should not attribute nasality in the vowel to coarticulation effects in any context, except nasal consonant. Randomized stimuli pairs containing oral, nasal and nasalized vowels in three different contexts (zero, nasal appendix, nasal consonant) were presented to BP listeners who had to judge which vowel was more nasal. Preliminary results demonstrate that BP listeners do not perceptually compensate for vowel nasality in any context, be it nasal appendix or nasal consonant.

1pSC6. Informational masking and the effects of differences in fundamental frequency and fundamental-frequency contour on phonetic integration in a formant ensemble. Robert J. Summers, Brian Roberts (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk), and Peter J. Bailey (Dept. of Psych., Univ. of York, York, United Kingdom)

This study explored the effects on intelligibility of across-formant differences in fundamental frequency (ΔF0). Sentence-length speech analogues were presented dichotically (left = F1 + F3; right = F2), either alone or—because competition usually reveals grouping cues most clearly—accompanied in the left ear by a competitor for F2 (F2C) that listeners must reject to optimize recognition. F2C was created by inverting the F2 frequency contour. In experiment 1, all left-ear formants shared the same constant F0 and ΔF0 for F2 was 0 or $\pm$4 semitones. In experiment 2, all left-ear formants shared the natural F0 contour and that for F2 was natural, constant, exaggerated, or inverted. Adding F2C lowered keyword scores, presumably because of informational masking. The results for experiment 1 were complicated by an asymmetry between the $\pm$4 semitone cases; this problem was avoided in experiment 2 because all four F0 contours had the same geometric mean. When the target formants were presented alone, scores were high and did not depend on the F0 contour for F2. F2C impact was greater when F2 had a different F0 contour from the other formants. This was a direct consequence of ΔF0; the F0 contour for F2 per se did not influence competitor impact. [Work supported by ESRC.]

1pSC7. Acoustic reduction, context, and inter-stimulus interval in cross-modal priming. Daniel Brenner and Benjamin V. Tucker (Linguist, Univ. of AB, 2-40 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, brenner@ualberta.ca)

Work such as Tucker (2011) and van de Ven *et al*. (2011) show that reduced acoustic contrasts can impede lexical access for a listener. Tucker (2011) also found preliminary evidence of a nonlinear (U-shaped) relationship between the degree of intensity dip in stops and lexical decision response times. The present talk summarizes results from a cross-modal identity priming experiment designed to explore this potential nonlinearity further. Utilizing three different inter-stimulus-intervals, trials were presented in which conversational words with variously reduced intervocalic stops served as auditory primes. Visual targets included the same word as the prime, words with a large degree of phonological overlap, and unrelated controls, as well as phonologically overlapping and non-overlapping pseudowords. Additionally, the auditory primes were presented with three degrees of surrounding context: isolation (the word only), phonetic context (including the vowels in neighboring syllables, providing primarily speech-rate information), or complete utterances. This talk explores the resulting picture of the relationship between reduction, context, and inter-stimulus-intervals to processing demand as evidenced in the response latencies. We then discuss the implications for models of perception and spoken word recognition.

1pSC8. American English schwa characteristics in the context of vowel reduction. Christina Kuo (Commun. Sci. and Disord., James Madison Univ., MSC4304, 801 Carrier Dr., Harrisonburg, VA 22807, kuocx@jmu.edu) and Gary Weismer (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI)

The mid-central unrounded vowel schwa has been understood as a vocalic produced with a uniform cross-sectional area of the vocal tract that is correlated with formant frequencies occupying the center of the acoustic vowel space. As such, schwas have also been associated with reduced vowels. Vowel reduction occurs in a variety of contexts and factors, and it is commonly described as centralization toward the middle of the vowel space or a migration of formant frequencies toward those associated with a tube of uniform cross-sectional areas. Nonetheless, little is known regarding the extent to which reduced vowels approximate schwa values. The purpose of this study was to establish a within-speaker reference of schwa in the context of vowel reduction. Two questions were addressed. First, do the acoustic characteristics of schwa correspond to the centroid of vowel space? Second, how do schwa values compare to those of reduced vowels? Formant frequencies of vowels in citation and connected speech tasks were compared to schwa productions for the same speaker. It was hypothesized that schwa can be characterized as a centroid region defined within individual speakers' vowel spaces. It was further hypothesized that reduced vowels in connected speech would closely approximate this centroid region.

1pSC9. Assessment of the starting point of the Lombard effect. Pasquale Bottalico, Ivano Ipsaro Passione, Simone Graetzer, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48910, pb@msu.edu)

The Lombard Effect is an involuntary tendency to raise the level of the voice with an increase in the background noise level. The aim of this study was to determine whether there is a starting point of the Lombard Effect in terms of background noise level (Ln).Twenty subjects were instructed to read a passage with the goal of being understood by a listener at 2m. A loudspeaker, also located at 2m, emitted pink noise. Ln varied in level from 25 dBA to 70 dBA in 5 dB increments. A regression model was fit with segmented relationships to the data, estimating the slopes and the change point(s) in the relationship between the response variable, within-subject normalised SPL, and the explanatory variable, Ln. One change point was identified at Ln 48.3 dBA. The rate of change in speech level below 48.3 dBA of noise was about 0.24 dB per dBA of noise while the rate of change above 48.3 dBA was nearly 0.5 dB per dBA of noise. Hence, a change point was identified in the speaker's adaptation to the background noise level, which could be said to mark the starting point of the Lombard Effect, as typically described.

1pSC10. Optimal time- and frequency-domain feature characterization for emotion recognition using electromyographic speech. Tejal Udhan and Shonda Bernadin (Elec. Eng., Florida State Univ., 2525 Pottsdamer St., Tallahassee, FL 32310, tu13b@my.fsu.edu)

In this paper, different approaches for characterizing emotions using pattern recognition in EMG signal are proposed. In time domain features, the variability index (VI) of EMG signal is used as a feature for pattern recognition. Fuzzy clustering toolbox in MATLAB is proposed as the tool for pattern recognition. VI is proven to be a good criterion for characterizing EMG signals, having a good account for amplitude and signal value at the given instant of time. However, there has been very little research on its contribution in identifying closely related emotions. Frequency domain features involving modified mean frequency (MMNF) and modified median frequency (MMDF) are used for different spectral characteristics of speech and kinematic signals. Since emotion recognition using EMG is a relatively new and innovative concept, the extraction of optimal features in the time-domain and frequency-domain that can be used for pattern recognition is a necessary initial step. This paper will explore the use of clustering techniques for optimal feature characterization of emotions.

1pSC11. Acoustic comparison of /t/ glottalization and phrasal creak. Marc Garellek and Scott Seyfarth (UCSD, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, mgarellek@ucsd.edu)

In American English, the presence of creaky voice can derive from distinct linguistic processes, including phrasal creak (prolonged irregular voicing, often at edges of prosodic phrases) and coda /t/ glottalization (when the alveolar closure for syllable-final /t/ is replaced by or produced simultaneously with glottal constriction). Garellek (2015) showed that listeners can differentiate words in phrasal creak from those with /t/ glottalization, which suggests that the creaky voice derived from phrasal creak and /t/ glottalization differ acoustically. To test this, we analyzed vowels preceding syllable-final /t/ in the Buckeye Corpus, which is comprised of audio recordings of spontaneous speech from 40 speakers of American English. Tokens were coded for presence of phrasal creak (prolonged irregular voicing extending beyond the target syllable) and /t/ glottalization (whether the /t/ was produced only with glottal constriction). Spectral measures of voice quality, including both harmonic and noise measures, were extracted automatically using VoiceSauce (Shue *et al.*, 2011), and discriminant analyses were performed. The results indicate several differences among non-creaky vowels, vowels before glottalized /t/, and vowels in phrasal creak. The discussion will focus on how different sub-categories of creaky voice can contrast with one another in a given language.

1pSC12. Acoustic classification of velar fricatives in Assamese. Charles Redmon (Linguist, Univ. of Kansas, 1541 Lilac Ln., Rm. 427, Lawrence, KS 66046, redmon@ku.edu)

Previous work on the acoustic discrimination of fricative contrasts has commonly studied categories which are not held to vary phonemically; instead the focus is placed on accounting for variation in the salience and stability of the acoustic cues mapping onto those categories (cf. Forrest *et al.*, 1988; Jongman *et al.*, 2000; McMurray and Jongman, 2011). The present study, like Jannedy *et al.* (2015), addresses a set of contrasts—the voiceless posterior obstruents /x, h, kh/ in Assamese—where /h/-voicing and /kh/-lenition processes interact with the discriminability of the velar fricative differentially according to positional (CV, VCV, and VC) and speaker (gender, dialect) variables. Results of principal components logistic regression (PCLR) models applied at both fixed and cumulative time windows over consonant and transition intervals are presented. Overall, /x/-/h/ discrimination was significantly more accurate in intervocalic position (96%) relative to word-initial position (84%). Data from intervocalic productions further suggest a significant dialectal difference in the /x/-/kh/ contrast, with discrimination poorer in Nalbari (68%) than in Jorhat (86%) data, in line with Sarma's (2012) analysis. Implications for the integration of lexical information by means of phonotactic priors derived from a corpus (Baker *et al.*, 2002) are also discussed.

1pSC13. Automatic classification of English fricatives using cepstral coefficients. Jason Lilley (Ctr. for Pediatric Auditory & Speech Sci., Nemours Biomedical Res., Wilmington, DE) and Laura Spinu (Anthropology, Univ. of Western ON, Social Sci. Ctr. Rm. 3314, Western University, London, ON N6A 3K7, Canada, lspinu@uwo.ca)

We use a classification tool previously tested on Romanian fricatives to categorize the front (non-sibilant) fricatives of English by place of articulation. Labiodental and interdental fricatives are difficult to distinguish acoustically, posing problems even for human perception. Prior classification work with English front fricatives has not been very successful with this contrast, with correct classification rates ranging from 40 to 60%. The feature set we use for coding the acoustic properties of the fricatives and their following vowels comprise the first six cepstral coefficients (c0–c5). The acoustic features are measured at 10-ms intervals across each segment; the measures obtained are then binned into three contiguous intervals for both the fricative and the vowel, representing the onset, steady state, and offset of each segment. The boundaries between regions are set by using a hidden Markov model to determine three internally uniform regions with respect to their acoustic properties. The mean value of each acoustic feature within each region is obtained; thus, each CV production yields six measurements per coefficient. We are testing this model on fricatives from the TIMIT corpus and classifying them using multinomial logistic regression models. While this investigation is underway, we expect higher correct classification rates than previous work.

1pSC14. The perception of breathy voice in Gujarati. Max Nelson (Linguist, Indiana Univ., 1021 East 3rd St. - Memorial Hall 322 E, Bloomington, IN 47405, maxnelso@umail.iu.edu)

Breathy voice, also referred to as murmur, is characterized by an increased open quotient and increased airflow. Acoustical correlates of breathiness include an increased spectral slope, most notably as measured by H1-H2 and H1-A3, as well as an increase in noise as measured by harmonics-to-noise ratio and cepstral peak prominence. Gujarati is relatively unique in that it contains phonemic breathiness in both consonants and vowels, creating meaningfully different C^{fi} V and CV.. sequences. Acoustic analysis of this contrast indicates that while the acoustic correlates of breathiness are consistent across these two sequences, there are differences in both the timing and magnitude of the cues of breathiness (Esposito and Khan 2012). Perceptual studies have indicated that speakers of languages with phonemic breathiness are capable of reliably differentiating breathy and modal sounds, but no study to date has addressed the perceptual salience of the timing and degree differences associated with breathiness in consonants versus vowels. The present study addresses the potential confusability of C^{fi} V and CV.. sequences in Gujarati through a series of perception tasks. Results indicate that there is an asymmetrical relationship between C^{fi} V and CV.. sequences that may be attributable to differences in the magnitude and timing of acoustic cues.

1pSC15. Using gated audiovisual speech perception to identify the temporal onset of coarticulation in production. Melissa A. Redford (Linguist Dept., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, redford@uoregon.edu), Sergei V. Bogdanov (Moonshadow Mobile, Eugene, OR), and Eric Vatikiotis-Bateson (Univ. of Br. Columbia, Vancouver, BC, Canada)

The goal of the present work was to develop a sensitive, non-invasive method for accurately assessing the temporal scope of coarticulation in speech production. Building on work in audiovisual speech perception (Munhall and Tohkura, 1998; Moradi *et al.*, 2013), we used a gating paradigm and human judges to identify the temporal onset of anticipatory lip rounding in simple SVO sentences produced by five adult females. The sentences were gated based on acoustic landmarks from the midpoint of the verb through to the midpoint of the object noun. Full sentences were also included. Judges were asked to decide whether the object noun rhymed with rounded "oop" or unrounded "ack." Results indicated an earlier correct identification of rounding in the audiovisual condition compared to the control, audio-only condition. The audiovisual judgments also provided greater temporal specificity regarding the onset of coarticulation than could obtained with acoustic measurement or with a video-based kinematic measure. Insofar as human judges appear to anticipate and even benefit from head movement in audiovisual speech tasks (Munhall *et al.*, 2004), our method should prove especially useful for measuring the temporal extent of coarticulation in younger, more rambunctious speakers.

1pSC16. Can audio-haptic speech be used to train better auditory speech perception? Josh Dorsi (Psych., UC Riverside, 34875, Winchester, NY 92596, jdors002@ucr.edu), Lawrence D. Rosenblum, James W. Dias, and Dana Ashkar (Psych., UC Riverside, Riverside, CA)

Training with audio-visual speech improves subsequent auditory speech perception more so than training with auditory-alone speech (e.g., Bernstein, *et al.*, 2013). What is the source of this bimodal training advantage? One explanation is that perceivers rely on learned bimodal associations. Alternatively, perceivers could be exploiting natural, amodal regularities available in both the auditory and visual signals. To test this question, multisensory training stimuli were tested for which observers had no bimodal associative experience. It is known that felt articulations —acquired by placing a hand on a speaker's face— can provide information for speech perception (see Trielle *et al.*, 2014, for a review). Importantly, these effects are found in participants with no prior experience perceiving speech through touch. If training with audio-haptic speech improves auditory speech perception, then this bimodal advantage cannot be due to learned associations but likely reflects sensitivity to amodal information. To test this hypothesis, participants either heard, or heard and felt, a speaker's speech. Participants subsequently identified words from a set of novel audio-alone sentences. Preliminary data indicate that audio-haptic speech training improves subsequent auditory-only perception more than audio-only training. These results challenge a learned association account for the bimodal training advantage.

1pSC17. Vowel discrimination asymmetry: The influence of token variability and lexical characteristics. Paula Garcia and Kanae Nishi (Hearing Res., Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68105, paula.garcia@boystown.org)

Listeners are more sensitive to changes in vowel quality when a peripheral vowel rather than a central vowel serves as deviant in an oddball paradigm (asymmetry in vowel discrimination). Schwartz *et al.* (2005) hypothesized that formant focalization in peripheral vowels make them perceptually more salient than central ones. Interestingly, however, a neurophysiological study (García and Froud, 2014) suggested that lexical familiarity may play a role in this perceptual asymmetry. This study investigated whether or not the asymmetry in vowel discrimination is mediated by the lexical characteristics of the stimulus words. Using an oddball paradigm, mismatch negativity (MMN) responses were elicited from adult monolingual American English listeners using naturally-produced American English vowel contrast /ɑ/-/ʌ/ in a real-word pair (Dock/Duck) and a non-word pair (Dop/Dupp). Listeners heard the pairs in two sessions. It was hypothesized that the MMN responses would be observed only when the peripheral vowel /ɑ/ in real words serves as deviant. Preliminary results suggest that MMN responses were observed for /ɑ/ but not for /ʌ/ regardless of the lexical status of the stimulus words. The implications of findings will be discussed in terms of the effects of lexical status of stimulus words and token variability in vowel discrimination.

1pSC18. Morphological effects on formant movement in spontaneous speech. Michelle Sims, Benjamin V. Tucker (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, mnsims@ualberta.ca), and Harald Baayen (Linguist, Univ. of Tübingen, Tübingen, Germany)

In the present study we investigate the role of morphology in the production of vowel formant movement and centralization. Our data consist of 74 monosyllabic irregular English verbs (6028 tokens) that differ between their present/past tense forms on a single vowel (e.g., *sing*/*sang*). For each vowel, we measured F1 and F2 contours and Euclidean distances from speakers' vowel space centres (measured at the vowel midpoint). A generalized additive model of the formant trajectories and a linear mixed effects regression model of the vowel centralization distances were created comparing two morphological predictors: verb tense (past or present), and paradigmatic support for the vowel. Paradigmatic support was measured using naïve discriminative learning as a metric which determined the association strength between a vowel and tense. A strong association with the past tense is evidence for greater paradigmatic support (i.e., it is more discriminating) while a vowel strongly associated with the present tense has low support. Our results indicate that the morphological predictors have an overall effect on formant movement (both in the past tense and with low paradigmatic support) and vowel centralization. However, the morphological effects differ between individual vowels.

1pSC19. The role of harmonic spectral structure in the cocktail party problem. Josh McDermott, Sara Popham, and Dana Boebinger (Brain and Cognit. Sci., MIT, 77 Massachusetts Ave., 46-4065, Cambridge, MA 02139, jhm@mit.edu)

Harmonicity is believed to provide an important acoustic grouping cue underlying sound segregation, though the mechanisms by which this occur, and its importance in real-world conditions, remain unclear. To test the role of harmonicity in the segregation of speech, we used a modified version of the STRAIGHT methodology for speech resynthesis to manipulate the fine-grained spectral structure of otherwise natural-sounding speech tokens. We then measured the ability of human listeners on two tasks: speech comprehension—of individual speech tokens in isolation, of speech in noise, and of speech presented concurrently with other tokens—and detection of a mistuned harmonic. We tested the importance of harmonicity in both tasks by jittering harmonic frequencies up or down by a small amount, rendering the speech inharmonic. We tested the importance of familiar spectral structure by deleting even-numbered harmonics. This latter manipulation left only the odd harmonics, a spectral pattern that, while harmonic, does not normally occur in natural sounds. Results indicate that harmonic speech excitation helps listeners to correctly group speech signals, aiding their comprehension amid concurrent talkers. However, the benefit does not generalize to unfamiliar harmonic spectra, consistent with the use of familiar spectral templates rather than harmonic structure per se.

1pSC20. Attentive tracking in human audition. Kevin Woods (Harvard Univ., 43 Vassar St., 46-4078, Cambridge, MA 02139, kwoods@mit.edu) and Josh McDermott (Massachusetts Inst. of Technol., Cambridge, MA)

Auditory scenes often contain multiple sound sources, but typically one is of particular interest and must be selected for further processing. This "cocktail party problem" is especially difficult when sources are similar and change over time (e.g., speakers of the same gender). To study this situation, we introduced a paradigm in which listeners attempt to follow one of two synthetic voices that vary randomly in several feature dimensions (e.g., f0, f1, and f2). Psychophysical results from this task suggest that human listeners employ a movable focus of attention that follows sources of interest through their feature space. Here we utilize this paradigm to probe the performance limits of attentive tracking. We also report that listeners can learn the abstract "shape" of an attentively tracked trajectory if it occurs repeatedly in a stimulus set. Knowledge of trajectory regularities appears to benefit the tracking of target sources, and demonstrates another way in which attentive tracking might aid our ability to hear under challenging listening conditions.

1pSC21. Methodology and technology for the polymodal allophonic speech transcription. Andrzej Czyzewski (Multimedia Systems Dept., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl), Tomasz E. Ciszewski (Faculty of Lang., Inst. of English, Univ. of Gdansk, Gdansk, pomorskie, Poland), and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Gdansk, Poland)

A method for automatic audiovisual transcription of speech employing: acoustic, electromagnetical articulography and visual speech representations is developed. It adopts a combining of audio and visual modalities, which provide a synergy effect in terms of speech recognition accuracy. To establish a robust solution, basic research concerning the relation between the allophonic variation of speech, i.e., the changes in the articulatory setting of speech organs for the same phoneme produced in different phonetic environments and the objective signal parameters (both audio and video) is carried out. The method is sensitive to minute allophonic detail as well as to accentual differences. It is shown that by using the analysis of video signals together with the acoustic signal, speech transcription can be performed more accurately and robustly than by using the acoustic modality alone. In particular, various features extracted from the visual signal are tested for their abilities to encode allophonic variations in pronunciation. New methods for modeling the accentual and allophonic variation of speech are developed. [Research sponsored by the Polish National Science Centre, Dec. No. 2015/17/B/ST6/01874.]

1pSC22. Selective adaptation of crossmodal speech information is not the result of higher-level stimulus associations. James W. Dias and Lawrence D. Rosenblum (Psych., Univ. of California, Riverside, 900 University Ave., Riverside, CA 92521, jdias001@ucr.edu)

Selective adaptation of speech information can change perceptual categorization of ambiguous phonetic information (e.g., Vroomen and Baart, 2012). The results of a number of studies suggest that selective adaptation may depend on sensory-specific information shared between the adaptor and test stimuli (e.g., Roberts and Summerfield, 1981; Saldaña and Rosenblum, 1994). For example, adaptation to heard syllables can change perception of heard syllables, but adaptation to lipread syllables has not been found to significantly change perception of heard syllables. The lack of crossmodal influence in selective speech adaptation is inconsistent with other phenomena suggesting crossmodal influences occur early in the speech process (e.g., Rosenblum, 2008). For the current investigation, a replication of an attempt to induce crossmodal speech adaptation again failed to yield significant effects. However, a meta-analysis including these results with past attempts (Roberts and Summerfield, 1981) revealed that small crossmodal speech adaptation effects are significant across studies. Next, an attempt to induce crossmodal speech adaptation employing a much larger group of new participants did yield significant crossmodal adaptation effects. A replication using text-stimuli failed to induce similar adaptation effects, suggesting that the observed crossmodal effects are not the result of higher-level stimulus associations. The theoretical implications of these smaller crossmodal effects will be discussed.

1pSC23. Vocal dosimeter devices and their uncertainty. Pasquale Bottalico, Ivano Ipsaro Passione, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48910, pb@msu.edu)

Vocal dosimeters may be used to characterize long-term voice production in typical daily activities. The reliability of four vocal dosimeter devices in the estimation of Sound Pressure Level (SPL) and fundamental frequency (Fo) was assessed. The devices evaluated were the Ambulatory Phonation Monitor (KayPENTAX), VoxLog (Sonvox), VocaLog (Griffin Laboratories), and Voice-Care (PR.O.VOICE). Twenty subjects were recorded in a sound booth by means of a head-mounted microphone while using each dosimeter. Subjects were asked to produce a sustained /a/ vowel and to read a text in three different voice styles (relaxed, normal, and raised). On the basis of the difference in the estimation of the parameters between the values acquired from the HMM and the dosimeters, the mean error, and the combined uncertainty were calculated. For the SPL, the dosimeter with the highest mean error was the APM, followed by the VoxLog, the VocaLog, and the Voice-Care. For Fo, the dosimeter with the highest mean error was the VoxLog, followed by the Voice-Care, and the APM (the VocaLog does not monitor Fo).

1pSC24. Categorical perception of visual speech information. Chase Weinholtz (Pitzer College, 1050 North Mills Ave., Pitzer College 835, Claremont, CA 91711, chaweitz@gmail.com) and James W. Dias (Psych., Pomona College, Riverside, CA)

Previous research suggests that many speech sounds are perceived in discrete categorical units. For example, perceivers typically identify acoustic stimuli from a continuum that ranges from one syllable to another (e.g., /va/-to-/ba/) exclusively as one syllable with a sharp change in which syllable the stimuli are identified as in the middle of the continuum. Further, pairs of continuum stimuli are typically easier to discriminate if they span this sharp change in stimulus identification. Walden *et al.* (1987) investigated whether visual (lipread) speech is perceived categorically using a continuum of digitally morphed mouths, but failed to find any strong evidence for categorical perception. In the current investigation, a technique developed by Baart and Vroomen (2010) was used to create a /va/-to-/ba/ continuum of visual test-stimuli. A video of a talker articulating /va/ was digitally superimposed over a video of the same talker articulating /ba/. Opacity of the /va/ video was then adjusted in 10% increments to create an 11-step continuum (from 0% to 100%). Identification and discrimination functions indicate that the visual continuum stimuli were categorically perceived as /va/ or /ba/. The results suggest that visual speech, like auditory speech, can be categorically perceived. Implications for gestural theories of speech will be discussed.

1pSC25. Effects of voice-onset time and talker variability on lexical access. Jieun Lee and Chao-Yang Lee (Ohio Univ., Gover Ctr. W218, Athens, OH 45701, jl976314@ohio.edu)

The effects of voice-onset time (VOT) and talker variability on accessing word meanings were investigated. Two short-term semantic priming experiments were used to evaluate if the magnitude of semantic priming would be reduced by these two types of variability. In experiment 1, a lexical decision task and a phoneme identification task were used to examine whether listeners' sensitivity to VOT affects their processing of a semantically related word. In experiment 2, a lexical decision task and a talker discrimination task were used to examine whether listeners' sensitivity to talker changes affects lexical access. The results showed that while listeners were sensitive to VOT and talker variability, neither affected the magnitude of semantic priming. Nonetheless, experiment 2 also showed that talker discrimination was easier when real word targets were presented, indicating that the word/non-word lexical status affected the processing of talker information. These findings, overall, suggest that there is no evidence that VOT and talker variability affect accessing word meanings.

1pSC26. Acoustics and perception of charisma in bilingual English-Spanish 2016 United States presidential election candidates. Rosario Signorello (Head and Neck Surgery, Univ. of California Los Angeles David Geffen School of Medicine, 31-24 Rehab Ctr., 1000 Veteran Ave, Los Angeles, CA 90095, rsignorello@ucla.edu)

The present work studies the acoustic voice profiles and the perception of charismatic traits of two bilingual candidates to the 2016 United States presidential election: Marco Rubio and Jeb Bush. Marco Rubio is a native bilingual English-Spanish speaker. Jeb Bush's native language is English and he is fluent in Spanish. The objective of the study is to investigate the differences and the similarities in speakers' acoustic voice profiles of the speakers addressing the audience in English versus in Spanish. The study also aims to assess the differences and the similarities in listeners' perception of speakers' charismatic traits while speaking in English vs. in Spanish. Monolingual English, monolingual Spanish, and bilingual English and Spanish listeners participating to the study assessed charisma traits perception through speakers' voice quality using the Multidimensional Adjective-based Scale of Charisma Perception (MASCharP, Signorello, 2014). Undergoing statistical analysis investigates the influence of individuals' native language in speakers' voice acoustic profiles and in listener's perception of charismatic traits through voice quality. [Work supported by NIH grant DC01797.]

1pSC27. Orthogonal interference of indexical information occurs even when phonetic contrasts are unambiguous across talkers. Ja Young Choi, Elly R. Hu, and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, jayoungc@bu.edu)

During speech perception, listeners maintain perceptual constancy despite highly variable associations between speech acoustics and abstract phonemic representations. Talker normalization facilitates speech processing by reducing the degrees of freedom for mapping encountered speech to phonemic representations. In our study, we investigated (i) whether talker normalization is obligatory even when there is no potential acoustic ambiguity between speech sounds and (ii) how the magnitude of potential ambiguity in acoustic-phonemic mappings affects talker normalization. We parametrically manipulated the phonetic similarity of words across three levels of difficulty for both vowels (easy: /bit/-/bot/, medium: /bɛt/-/bʌt/, hard: /bot/-/but/) and consonants (easy: /baɪ/-/saɪ/, medium: /baɪ/-/taɪ/, hard: /baɪ/-/paɪ/). In a series of speeded classification paradigms, listeners identified these words spoken by single or mixed talkers. Auditory word categorization was always slower in mixed-talker conditions than in single-talker conditions, even when there was no potential acoustic ambiguity between the target sounds. Furthermore, the processing cost imposed by mixed talkers was greatest when words were most similar phonetically. The results demonstrate (i) that talker normalization is obligatory in speech processing even when sounds are wholly acoustically distinct and (ii) that the facilitatory effect of talker normalization varies as a function of the potential ambiguity of acoustic-phonemic mapping.

1pSC28. Perception of voice onset time in Singapore English. Priscilla Z. Shin (Dept. of Linguist & Anthropology, Univ. of Arizona, P.O. Box 210025, Tucson, AZ 85712, pzl2@email.arizona.edu)

This study investigates how voice onset time (VOT) is perceived in Singapore English (SgE). SgE research acknowledges that Singaporeans linguistically accommodate their listeners—varying between using Standard Singapore English (SSE) or Singlish (colloquial SgE) depending on addressee. Liu 2011 (n.p.) suggests these linguistic accommodations are reflected in VOT production, finding Singaporeans to produce more short-lag VOT (indicative of Singlish) when speaking to other Singaporeans and

longer VOT (indicative of SSE) when speaking to Americans. Though these are sub-categorical shifts, they demonstrate the utility of VOT in speakers' efforts when accommodating listeners. The current study tests whether VOT is also available as a salient and social perceptual cue. Three listening experiments investigate (1) Singaporeans' baseline categorical perception of a /ba/-/pa/ continuum, (2) whether this baseline is affected when heard with socially salient pictures (i.e., linguistic stereotypes—an *ah lian* or a business woman), and (3) whether dialectal carrier phrases—whether stimuli are presented within a SSE or Singlish phrase—affects perception of the same stimuli. Preliminary results show listener age and educational background, which allude to an effect of dialectal experience (Sumner and Samuel 2009), play important roles in the perception of VOT.

1pSC29. Extrinsic talker normalization via rapid accumulation of talker-specific phonetic detail. Tyler Perrachione and Ja Young Choi (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, tkp@bu.edu)

Talker normalization makes speech processing more efficient by reducing the degrees of freedom between the highly variable acoustic realization of speech and its abstract mental representation. However, it is unknown how processing gains afforded by talker normalization depend on the amount of talker-specific information available in the preceding speech context. We explored the timecourse of extrinsic talker normalization in a series of speeded classification tasks that parametrically varied the amount of talker-specific phonetic detail available in the preceding speech context. In single- and mixed-talker conditions, listeners identified a target word (either “boot” or “boat”) preceded by either no speech context, a short carrier phrase (“It's a...”), or a long carrier phrase (“I owe you a...”) spoken by the same talker as the target word. Listeners' word identification was always slower in the mixed-talker condition than the single-talker one. However, the amount of interference decreased as the amount of preceding speech context increased: the interference effect of the mixed-talker condition was greatest with no carrier, less with a short carrier, and least in the long-carrier condition. These results reveal how extrinsic talker normalization facilitates speech processing via rapid accumulation of talker-specific detail from the speech context.

MONDAY AFTERNOON, 23 MAY 2016 SALON I, 1:30 P.M. TO 3:20 P.M.

Session 1pSP

Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics: Small Unmanned Aerial Vehicle (UAV) Detection, Tracking, and Classification

R. Lee Culver, Cochair
ARL, Penn State University, PO Box 30, State College, PA 16804

Geoffrey H. Goldman, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

1:30

1pSP1. Overview of civilian counter drone systems. Christoph Borel-Donohue (SEDD, US Army Res. Lab., 2800 Powder Mill Rd, Bldg, 202, Rm. 3F068, Adelphi, MD 20783, cborelc@gmail.com)

Unmanned aerial vehicles (UAV) commonly known as drones pose a significant threat to civilian and military personnel. This talk will explore several commercially available counter UAV systems developed for the civilian market. The systems discussed make use of electro-optic, acoustic, and radio frequency signatures to detect the presence of UAVs. The talk will also show some examples of acoustic, electro-optic, and radio frequency signatures measured on a few commercial drones.

1:50

1pSP2. Localization and separation of acoustic sources by using a 2.5-dimensional circular microphone array. Mingsian R. Bai and Chang-Sheng Lai (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

Circular microphone arrays (CMA) are preferred over more complex spherical microphone arrays (CMA) in the context of some audio applications because azimuth angles of spatial sound are considered more important than the elevation angles in those scenarios. However, CMA treats only two-dimensional sound fields without considering elevation angles, which can be a limitation in spatial audio rendering. This paper proposes a 2.5-dimensional (2.5D) CMA that consists of a baffled CMA and a vertical linear array on the top. In the localization stage, two minimum-variance-distortionless-response (MVDR) beamformers are applied to the circular array and the linear array, respectively. The product of the identified angular patterns yields the direction of arrival (DOA). In the separation stage, Tikhonov Regularization (TIKR) and Compressive Sensing (CS) are employed to extract the source signal amplitudes from the output signals from two arrays. The extracted signals are further processed by using an adaptive correlator to produce the source signal with improved quality. To validate the 2.5D CMA experimentally, a three-dimensionally printed circular array comprised of a 24-micro-electro-mechanical system (MEMS) microphone circular array and an 8-MEMS microphone linear array is constructed for localization and separation for sound sources.

2:10

1pSP3. Localization of sources on the ground with airborne acoustic platforms in a refractive, turbulent atmosphere. Vladimir E. Ostashev, D. K. Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@noaa.gov), Sylvain Cheinet (French-German Res. Inst. of Saint-Louis, Saint-Louis, France), Christian Reiff, Sandra L. Collier, Jonn Noble, and David Ligon (U.S. Army Res. Lab., Adelphi, MD)

Acoustic sensors are being employed on airborne platforms for localization of sources on the ground. In this presentation, the impact of atmospheric turbulence and refraction of sound signals due to the vertical profiles of temperature and wind velocity on source localization are overviewed. The variance of the angle of arrival of sound signals in a turbulent atmosphere is analyzed. A theory is presented which accounts for refraction corrections in source localization with elevated sensor arrays. The theory expresses the horizontal, azimuthal, and elevation refraction corrections in terms of the vertical profiles of temperature and wind velocity and the sensors' coordinates. The theory is applied to the results of a comprehensive experiment in source localization with an aerostat-mounted acoustic sensor array at Yuma Proving Ground (YPG). Long term atmospheric data sets from weather modeling systems are used for a climatological assessment of the refraction corrections and localization errors. Due to reciprocity in sound propagation, the results presented are also pertinent to localization of unmanned aerial vehicles (UAVs) with ground-based acoustic sensors.

2:30

1pSP4. Acoustic detection and tracking of UAS targets. Christian G. Reiff (Signal and Image Processing, Army Res. Lab., 2800 Powder Mill Rd., RDRL-SES-P, Adelphi, MD 20783, christian.g.reiff.civ@mail.mil)

Acoustic detection using a Minimum Variance Distortionless Response (MVDR) method and tracking of Unmanned Aircraft System (UAS) targets with the MVDR results over time will be presented. The acoustic environment was complex with surf noise, wind, road traffic, air traffic, and aircraft hangar noise. The targets were gas and electric fixed wing UAS as well as electric multi-rotor quadcopters. Scenarios with the UAS target leaving, incoming, and with multiple simultaneous targets are addressed. Tracking results are evaluated through comparison with GPS based UAS ground truth tracks.

Contributed Papers

2:50

1pSP5. The effect of Doppler rate on small aperture bearing estimation. David C. Swanson and Robert L. Culver (Penn State ARL, 222E ARL Bldg., PO Box 30, State College, PA 16804, dcs5@psu.edu)

Passive estimation of the angle of arrival using a small aperture array outdoors can be adversely effected by high rates of frequency chirping. This is caused by either rapid change in source frequency and/or rapid change in Doppler frequency due to the motion of the source. Consider two microphone signals simultaneously sampled and analyzed for phase difference via FFT. The time difference of arrival across the small aperture array is small compared to the FFT buffer size, so there are slightly different starting and stopping frequencies for each microphone during the FFT input buffer record. However, the chirp (rate of change in Doppler or source frequency) leads to a significant phase error between the two microphones. An upwards chirp moves the estimated bearing erroneously towards 0 degrees while a downward chirp moves the estimated toward 180 degrees. For tracking aircraft from the ground it is not unusual to experience high rates of change in Doppler frequency near the closest point of approach. This paper shows the extent of the error from practical Doppler and source chirps and how to correct for the bearing error.

3:05

1pSP6. Frequency-dependent beam pattern of acoustic arrays for unmanned aerial vehicle tracking. R. Lee Culver (ARL, Penn State Univ., PO Box 30, State College, PA 16804, rlc5@psu.edu)

There are practical constraints on the geometry and construction of acoustic arrays which are designed to detect small unmanned aerial vehicles (UAVs). The constraints depend to a large degree on the environment in which the array must operate. For example, an acoustic array which provides UAV detection to protect a public stadium is likely to differ considerably from one which soldiers can carry into the field and deploy to protect their position. The former can use almost unlimited power, be very large and be composed of many elements, while the latter will probably run on batteries or solar and must be man-portable. These different designs provide different performance as measured by detection range, array gain, and angular resolution, for example. This talk presents the performance of several common array configuration over the band of frequencies in which passive AUV detection at tactically useful ranges is possible.

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

C. F. Gaumond, Chair ASC S2
14809 Reserve Road, Accokeek, MD 20607

J. T. Nelson, Vice Chair ASC S2
Wilson Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

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. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, mechanical vibration, shock and condition monitoring, and four of its subcommittees, take note that the meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2016.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance, and comfort.

Payment of an additional registration is required to attend.

MONDAY EVENING, 23 MAY 2016 SALON B/C, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Acoustic Metamaterials: From Theory to Practice

Christina J. Naify, Chair
Acoustics, Naval Research Lab., 4555 Overlook Ave. SW, Washington, DC 20375

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Acoustic metamaterials: From theory to practice. Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854-8058, norris@rutgers.edu)

Why have acoustic metamaterials appeared on the scene in the last few years, and what are they anyway? The original defining property of a metamaterial is that it achieves effects not found in nature, with the more effects the better. The acoustic cloak, justifiably the best known example, redirects sounds around the cloaked object regardless of the incident acoustic wave. The common theme of the cloak and of the many other devices that have been proposed is defiance of the intuitive laws of acoustical physics as much as possible. This requires strange concepts such as acoustic anisotropy, allowing sound to have speed that depends on direction, and transformation acoustics which is the basis for acoustic cloaking. Periodic structures, aka phononic crystals, play a central role in practical applications. The tutorial will provide a comprehensive introduction to these ideas without the need for detailed mathematics. While the field of metamaterials originated in electromagnetics before it migrated over to acoustics, there are fundamental differences between the two fields, the most obvious arising from the fact the acoustic wave speeds are not strictly limited, a distinction which leads to profound consequences. Acoustic wavelengths are also typically orders of magnitude larger than optical wavelengths, meters versus microns, which makes the acoustic problems easier, in principle. This practical advantage is perhaps the main reason for the intense interest in the field by funding agencies and researchers alike.

Session 2aAAa

Architectural Acoustics and Noise: Forensic Studies from Noise Identification to Solutions in the Built Environment

Kenneth Cunefare, Cochair
Georgia Tech, Mechanical Engineering, Atlanta, GA 30332-0405

Kenric D. Van Wyk, Cochair
Acoustics By Design, Inc., 124 Fulton Street East, Second Floor, Grand Rapids, MI 49503

Chair's Introduction—8:00

Invited Papers

8:05

2aAAa1. What's that sound? Probe tones and poltergeists: Two case studies in diagnosing noise sources. Kenneth A. Cunefare (Georgia Tech/Arpeggio Acoust. Consulting, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

Case #1: Following construction of a new hospital building within a larger campus, residents began to complain about droning noises in the surrounding neighborhood. Blame was laid on HVAC chillers and an MRI chiller installed on exposed roof-tops of the building. However, there were numerous other chillers on buildings on the campus that were not blamed; a test using probe tones eliminated the hospital building as the source impacting the community. Case #2: A homeowner complained of intermittent groaning and moaning noises in the home (poltergeist-like). Interviews revealed that the noises occurred essentially randomly; while the homeowner correlated occurrences to whether or not it had recently rained, the correlation was weak. Additionally, the homeowner had found that opening a water tap would cause the sound to fade away; this was suggestive, but since the sound faded away in a short time even without opening the valve, it was not definitive. Complicating the diagnosis was the intermittency of the noise (weeks could pass between events), and its short duration. A faulty GFI threw a red-herring into the mix, but ultimately the problem was solved by replacing the household water pressure regulator.

8:25

2aAAa2. Trains, transformers, and tinnitus: Case studies in hard-to-detect sounds. Arno S. Bommer and Edgar Olvera (CSTI Acoust., 16155 Park Row, Ste. 150, Houston, TX 77084, arno@cstiacoustics.com)

It's a phone call many acoustical consultants dread: someone driven crazy by an annoying sound. Sometimes, the source is unknown. Sometimes it's a nearby industry or antagonistic neighbor. Some sounds are intermittent; some are steady. Some are high frequency; others are low and are described as vibration. Most consultants are greatly relieved if they can hear and measure the sound themselves. But the source can still be elusive, especially if the sound is very faint and if there are multiple potential sources. More often than not, the consultant can hear and measure nothing, which gets blamed on his poor hearing and shoddy equipment. A discussion of tinnitus is often received as an accusation of "hearing things." But is tinnitus (or a similar condition) the only possible explanation? Why is the sound heard only at home? What triggered the person to start hearing it? The person may be especially sensitive, but to what? What advice is useful? The authors will present several case histories where community sounds were ultimately identified. For those sounds not identified, the author will discuss a number of possible explanations along with recommendations for the consultant and the person hearing the sound.

8:45

2aAAa3. The singing parking garage: Wind induced Helmholtz resonation of perforated metal panels. Erik Miller-Klein (SSA Acoust., LLP, 222 Etruria St., Ste. 100, Seattle, WA 98109, erik@ssaacoustics.com)

An above ground parking garage with perforated metal facade experienced high amplitude sound when northwest winds exceeded 10 mph around this building. The noise only occurred under very specific weather conditions, which required on-call noise and vibration measurements to identify the cause and feasible solution. The noise was traced to the Helmholtz resonator effect caused by the perforated metal panel holes and the air cavity associated with the headlight screens behind these perforated facade panels. Through our on-site investigation and analysis, the root cause was identified and a low cost remediation plan was designed and implemented.

9:05

2aAAa4. Methods for the identification of intermittent and unpredictable noises. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

Acoustical consultants are often asked to identify noises that are intermittent and unpredictable, such that attended observation is impractical. This paper presents methods for recording and identifying such events using unattended long-term recordings of sound and vibration, and for efficiently analyzing the voluminous data that can accumulate during such studies.

9:25

2aAAa5. Room acoustic design, measurement, and simulation techniques to reduce hospital noises within patients' environment. Mojtaba Navvab (Architecture, Univ. of Michigan, 2000 Bonisteel Blvd., Art and Architecture Bldg., Ann Arbor, MI 48109-2069, moji@umich.edu)

Hospitalized patients and clinical staff identify noise as a major stressor. Environmental hospital noise raises ambient noise levels significantly above ideal levels. Options to reduce hospital noise include methods to modify noise, such as closing doors, adjusting hospital equipment, hospital personnel behaviors, and clinical alarms such as wireless communication devices for staff and future "smart" algorithms for patient-specific alarm thresholds. This study utilizes the beamforming techniques to localize and examines basic characteristic of sound field through measurements and simulation of noise in an environment that affects the patient's audio perception, and collects evidence for impact of noise intensity on patient hearing toward reduction of the noise sources in hospital rooms. The profiles of noise levels for their time and frequency domain as being reflected, absorbed, or refracted within typical patient's room are determined. The difference in applications of various architectural solutions to mitigate noise in hospitals within patient environment is estimated. The results contribute toward architectural room acoustic design solution(s) to reduce acoustic disruption of sleep or rest with adequate information and reproducible data to accelerate design decision-making process while providing practical solutions such as geometric modification to room architectural elements as compared to theoretical offerings by the research community.

9:45

2aAAa6. Localization of plumbing noise in a multistory building using an acoustic camera. Marek Kovacik (Scantek, Inc., 6430 Dobbin Rd., Ste. C, Columbia, MD 21045, m.kovacik@scantekinc.com) and Jørgen Grythe (Norsonic, Lierskogen, Norway)

Noise generated by plumbing pipes inside a multi-family dwelling can be a source of great annoyance on its residents. Corrective action of such noise begins with identification of the noise source followed by appropriate noise reduction measures. Source identification becomes more challenging when documentation on the plumbing design is missing or when the noise level is so low that it cannot be localized using practical methods. This paper describes a similar problem where low level noise generated by sanitary pipes is heard inside an apartment building. As a localization tool, an acoustic camera utilizing beam-forming technology is used to identify the noise location inside walls between two adjacent apartment units. Findings are visually presented and discussed.

10:05–10:20 Break

Contributed Papers

10:20

2aAAa7. Gunshot recordings from a criminal incident: Who shot first? Robert C. Maher (Elec. & Comput. Eng., Montana State Univ., 610 Cobleigh Hall, PO Box 173780, Bozeman, MT 59717-3780, rob.maher@montana.edu)

Audio forensic examination for law enforcement and criminal justice investigations increasingly involves audiovisual recordings from dashboard camera systems, bystander smart phones, body cameras worn by police officers, and even by cameras built into TASER™ devices. If the camera is pointing in an appropriate direction the details of the incident may be found in the recorded video. However, if the camera's field of view is limited, it may still be possible to evaluate the circumstances of interest by examining the sounds captured by the recording device's microphone. This paper presents audio examples in which the forensic examiner must attempt to address questions such as: How many gunshots took place? What types of firearms were involved? Who shot first? Audio examples are presented to demonstrate the solutions—and mysteries—found in several real world cases.

10:35

2aAAa8. Noise identification and ongoing research and monitoring for Red Rocks Amphitheatre. Ted Pyper and Matt Whitney (K2 | Consultants in Audio, Video and Acoust., 4900 Pearl East Cir., Ste. 201E, Boulder, CO 80301, ted@k2audio.com)

Beginning in 2013, a noise survey was initiated by the famed Red Rocks Amphitheatre, outside Denver, CO, to respond to noise concerns from nearby communities. Complaints focused on excessive bass frequencies originating from the outdoor venue. Some neighbors even complained of physical vibrations in their residences from amplified music over a mile away. The survey was executed to document the level differences between the sources in the venue and the measured levels in the neighborhoods. Vibration levels were also measured, in addition to airborne noise levels, to verify and diagnose the specific spectrum associated with the complaints. Remedial treatments were studied, and policies have been put in place to monitor and control the overall levels within the venue. Ongoing monitoring and research have evolved over the last several years to better address the concerns of neighbors, while preserving the concert experience within the venue.

10:50

2aAAa9. Shot in the dark: Physical and psychophysical limitations on shooter identification from earwitness testimony. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

While human listeners have a remarkable ability to determine the cause of a sound and the location of the sound source using only their sense of hearing, this ability is subject to errors and lack of certainty due to factors that are grounded in well-understood phenomena in physics and psychoacoustics. In this talk, the author will describe his experience serving as an expert witness for the defense in a criminal trial concerning an incident that took place in a dark outdoor location in which a shot was allegedly fired by one of two suspects. The defendant was alleged to be the shooter based upon earwitness testimony regarding the direction from which the shot was heard to have originated. Three key questions brought out in the forensic analysis performed for the trial will be addressed from the perspective of signal-detection theory considering both physical factors (e.g., outdoor blast and shockwave propagation) and psychophysical factors (e.g., auditory detection and classification and spatial-hearing acuity): (1) Was the gunshot audible? (2) Was the sound that was heard identifiable as a gunshot? (3) With what accuracy and precision could the location of the gunshot be determined?

Session 2aAAb

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition

Andrew N. Miller, Cochair
Bai, LLC, 4006 Speedway, Austin, TX 78758

Michael Ermann, Cochair
Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205

David S. Woolworth, Cochair
Oxford Acoustics, 356 CR 102, Oxford, MS 38655

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund and the National Council of Acoustical Consultants are sponsoring the 2016 Student Design Competition that will be professionally judged at this meeting.

A center for the arts has decided to open a multi-purpose facility. It will include an auditorium with a balcony and platform (stage) to be used as a meeting space and as a music performance chamber. The facility will also include a green room to be used as a meeting room and for music rehearsal. Music performed and rehearsed in this facility will be chamber ensembles and soloists. Instrumentation will be exclusively classical, western orchestral.

The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of USD$1250 will be made to the submitter(s) of the design judged "first honors." Four awards of USD$700 each will be made to the submitters of four entries judged "commendation."

TUESDAY MORNING, 24 MAY 2016 SALON I, 8:00 A.M. TO 10:10 A.M.

Session 2aAO

Acoustical Oceanography and Signal Processing in Acoustics: Acoustic Consistency of Ocean Models

Timothy Duda, Cochair
Woods Hole Oceanographic Institution, WHOI AOPE Dept. MS 11, Woods Hole, MA 02543

Bruce Cornuelle, Cochair
UCSD-Scripps Institution of Oceanography, 9500 Gilman Drive, Dept. 0230, La Jolla, CA 92093-0230

Chair's Introduction—8:00

Invited Papers

8:05

2aAO1. Multiscale ocean prediction and uncertainty estimation for acoustic studies. Pierre F. Lermusiaux (MIT, 77 Mass Ave., Cambridge, MA 02139, pierrel@mit.edu)

We discuss the prediction and estimation of multiscale ocean fields and their probability density distribution for acoustic studies. In high-fidelity multi-resolution simulations, the probability density function of the full ocean state is predicted and estimated, combining the governing equations with observations. Dynamically balanced stochastic forcing are included, so as to represent effects of sub-grid-scales not resolved by the deterministic model equations. The results are stochastic partial differential equations that allow to capture both deterministic effects (advection, Coriolis, etc.) and statistical effects (smaller-scale turbulence, internal wave variability, etc.) on the environment. With this modeling and data assimilation, accurate estimates of the probability density functions (pdf) of oceanographic variability are possible. They become inputs to end-to-end oceanographic-seabed-acoustic-sonar dynamical systems. This end-to-end uncertainty quantification approach is described, evaluated, and illustrated in several simulations and ocean regions.

8:30

2aAO2. Use of ocean model forecasts during acoustic experiments and naval missions. Kevin D. Heaney (OASIS, Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

Ocean acoustic propagation is critically dependent upon the spatial (and to a lesser extent the temporal) scales of the ocean. With the advent of high-fidelity data-assimilative ocean models, forecasts are available that can provide an improved level of confidence for acoustic propagation modeling in support of sea-tests and naval exercises. In this paper, a set of experiments will be presented where ocean model forecasts were used within an acoustic model to help inform the experiment planning. Tests include shallow water tests (Key West), continental shelf-break environments (Shallow Water 2006, Quantifying and Predicting Uncertainty) and deep water environments (Philippine sea 2009/2010). Models used include the Navy Coastal Ocean Model (NCOM), the MSEAS model (MIT, Pierre Lermusiaux), the ECCO2 model state estimation and the Scripps Institution of Oceanography version of the ROMS (Bruce Cornuelle) model run. *In-situ* assimilation of acoustic signals was performed for geo-acoustic information (in shallow water) and not for ocean model forecast updates. Our conclusion for over 10 years of use of models in real-time experiments is that ocean model forecasting does provide useful information for mission planning and experiment design, particularly when combined with on-site measurements.

Contributed Papers

8:55

2aAO3. Passive ocean acoustic tomography using ships as sources of opportunity recorded on an irregularly spaced free-floating array: A feasibility study. Jacquelyn S. Kubicko, Christopher M. Verlinden (U.S. Coast Guard Acad., New London, CT 06320, jkubicko@gmail.com), Brendan V. Nichols, and Karim G. Sabra (Mech. Eng. Dept., Georgia Inst. of Technol., Atlanta, GA)

This presentation investigates the practical feasibility of using an adaptive volumetric array, such as a series of free-floating buoys with suspended hydrophones, which record ships as acoustic sources of opportunity in coastal waters for performing acoustic thermometry or other environmental inversions in near-shore environments in a totally passive manner. Ships are tracked using the Automatic Identification System (AIS). Numerical simulations using a standard normal mode propagation model were first used to test limitations of the proposed approach with respect to frequency band, drifting receiver configuration, signal to noise ratio, precision, and accuracy of the inversion results, along with sensitivity to environmental and position mismatch. Performance predictions using this model are compared with experimental results using at-sea data collected off the coast of New London, CT, in Long Island Sound during August of 2015.

9:10

2aAO4. A reformulation of the ΛΦ diagram for the prediction of ocean acoustic wave propagation regimes. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

The ΛΦ diagram was a tool introduced in the late 1970s to predict ocean acoustic fluctuation regimes termed unsaturated, partially saturated, and fully saturated, where internal wave sound-speed fluctuations play a dominant role. The ΛΦ parameters reflect, respectively, the strength of diffraction and the root mean square phase fluctuation along a ray path. New oceanographic knowledge of the small scale part of the internal wave spectrum, and high angle Fresnel zone formulations now allow a more stable and accurate calculation of these parameters. An empirical relation between the variance of log-intensity and ΛΦ provides a more accurate border between the unsaturated regime and stronger fluctuations. The new diagram is consistent with six short-range, deep-water experiments in the Pacific, Atlantic, and Arctic oceans with frequencies ranging from 75 to 16000 Hz. The utility of the ΛΦ diagram is that it provides one of the few means to inter-compare experiments at different geographic locations, different frequencies, and ranges.

9:25

2aAO5. Ray travel times in the northern Philippine Sea circulation from state estimates. Bruce Cornuelle, Ganesh Gopalakrishnan, Matthew Mazloff, Peter F. Worcester, and Matthew A. Dzieciuch (UCSD-Scripps Inst. of Oceanogr., 9500 Gilman Dr., Dept. 0230, La Jolla, CA 92093-0230, bcornuelle@ucsd.edu)

The North Pacific Acoustic Laboratory (NPAL) Philippine Sea experiment deployed a variety of instruments in the northern Philippine Sea during April 2010 through April 2011, including six acoustic transceivers for reciprocal measurements of travel times between instruments. Five transceivers were moored in a pentagon approximately 660 km in diameter with the sixth transceiver moored in the center. Observed travel-time time series were compared with travel times computed from ocean state estimates made using an eddy-active regional implementation of the MITgcm that were constrained by satellite sea surface height and sea surface temperature observations and by temperature and salinity profiles from Argo, CTDs, and XBTs but not by the acoustic data. The similarities cross-validate the state estimates, while the differences provide a simple estimate of the novel information present in the travel times. Smoothness in the modeled sound speed field had a large effect on the ability to find eigenrays. The ocean state estimates were then re-computed to fit the acoustic travel times. The state estimate was able to match the travel times within their error bars and did not significantly increase the misfits with the other observations.

9:40

2aAO6. The integrated ocean dynamics and acoustics project. Timothy F. Duda, James F. Lynch, Ying-Tsong Lin, Weifeng G. Zhang, Karl R. Helfrich (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), Harry L. Swinney (Univ. of Texas at Austin, Austin, TX), John Wilkin (Rutgers Univ., NB, NJ), Pierre F. Lermusiaux, Nicholas C. Makris, Dick Y. Yue (Massachusetts Inst. of Technol., Cambridge, MA), Mohsen Badiey (Univ. of Delaware, Newark, DE), William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY), Jon M. Collis (Colorado School of Mines, Golden, CO), John A. Colosi (Naval Postgrad. School, Monterey, CA), Steven M. Jachec (U. S. Naval Acad., Annapolis, MD), Arthur E. Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA), Lin Wan (Univ. of Delaware, Newark, DE), Yuming Liu (Massachusetts Inst. of Technol., Cambridge, MA), Matthew S. Paoletti (Univ. of Texas at Austin, Austin, TX), Zheng Gong, Patrick J. Haley (Massachusetts Inst. of Technol., Cambridge, MA), Likun Zhang (Univ. of Texas at Austin, Austin, TX), Kaustubha Raghukumar (Naval Postgrad. School, Monterey, CA), and Michael R. Allshouse (Univ. of Texas at Austin, Austin, TX)

The goal of timely and accurate acoustics modeling in the ocean depends on accurate environmental input information. Acoustic propagation modeling has improved to the point of possibly being ahead of ocean dynamical modeling from the standpoint that some significant ocean features having strong acoustic effects are not faithfully reproduced in many models, particularly data-driven ocean models. This in part stems from the fact that ocean models have developed with other goals in mind, but computational limitations also play a role. The Integrated Ocean Dynamics and Acoustics (IODA) MURI project has as its goals improving ocean models, and also making continued improvements to acoustic models, for the purpose of advancing ocean acoustic modeling and prediction capabilities. Two major focuses are improved internal tide forecasting and improved nonlinear internal wave forecasting, which require pushing the state of the art in data-constrained mesoscale feature modeling as well as developing specialized high-resolution tools. Results are reported on efforts to evaluate internal-tide accuracy in data-constrained models, to insert typically unresolved nonlinear internal waves with nonhydrostatic pressure dynamics into these models, and to make acoustical condition forecasts within three-dimensional operational volumes filled with internal waves.

9:55

2aAO7. Estimating sea-water volume attenuation coefficients from mid-frequency, deep-water experiments in the Atlantic and Pacific. Jeffery D. Tippmann, Jit Sarkar, Chris Verlinden, William S. Hodgkiss, and William A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, jtippman@ucsd.edu)

A collection of mid-frequency deep-water experiments were performed off the U.S. Pacific coast and in the northeastern Atlantic using a short vertical array cut for 7.5 kHz. Previously presented analysis of a Pacific experiments have shown agreement between experimental estimates of attenuation coefficients to the decades old attenuation models. We present the comparison of experimental attenuation coefficient estimates from additional Pacific and Atlantic data and discuss the methodology for inverting for depth-dependent ocean properties using acoustic amplitude measurements. The data and methodology are also used to explore the feasibility of ocean acoustic attenuation tomography for determining the individual chemical attenuation coefficient terms strictly from *in situ* measurements.

TUESDAY MORNING, 24 MAY 2016 — SNOWBIRD/BRIGHTON, 8:00 A.M. TO 9:45 A.M.

Session 2aBAa

Biomedical Acoustics: Acoustic Radiation Force and Elastography

Armen Sarvazyan, Chair
Artann Laboratories, 1753 Linvale-Harbourton Rd., Lambertville, NJ 08530

Contributed Papers

8:00

2aBAa1. Toward validation of shear wave elastography using vibration rheometry in soft gels. Sanjay S. Yengul (Mech. Eng., Boston Univ., 216 Saint Paul St. #201, Brookline, MA 02446, syengul@bwh.harvard.edu), Bruno Madore (Radiology, Brigham and Women's Hospital, Harvard Univ., Boston, MA), and Paul E. Barbone (Mech. Eng., Boston Univ., Brookline, MA)

Shear wave propagation is at the heart of several elastography methods designed to quantify the elastic properties of living tissues. These differ in excitation mechanism, imaging modality, volume of tissue being sampled, and the diagnostic output [Parker *et al.* PMB, 2011]. Some of these techniques are already available to clinicians as diagnostic tools for the identification of pathology, particularly in the diagnosis of liver fibrosis. However, these techniques work in different parameter regimes, making it difficult to quantitatively compare different elastography modalities. As a result, the critical cutoffs for fibrosis staging are system dependent [Ferraioli *et al.* WFUMB, 2015]. We present progress toward a framework to allow quantitative comparison of different viscoelastic measurement techniques, including shear wave based elastography (SWE) methods (e.g., acoustic radiation force based SWE and magnetic resonance elastography), and a novel vibration rheometry of gel phantoms. In so doing, we seek to account for frequency and temperature dependence of material properties, as well as control for other confounding factors such as water content, material aging, boundary conditions, awkward sample geometries, and batch-to-batch gel variations.

8:15

2aBAa2. Potential biomedical applications of non-dissipative acoustic radiation force. Armen Sarvazyan (Artann Labs., 1753 Linvale-Harbourton Rd., Lambertville, NJ 08530, armen@artannlabs.com) and Sergey Tsyuryupa (Artann Labs., Trenton, NJ)

Acoustic radiation force (ARF) has been used for many medical applications, the most notable of which is measuring the stiffness of soft tissues. The ARF used in these applications results from the dissipation of acoustic energy due to scattering and absorption of sound and accordingly may be called dissipative ARF. Another source of ARF, which is not related to dissipation of acoustic energy in medium, is variation in acoustic energy density due to gradients of compressional wave speeds in the medium. The effects of this non-dissipative ARF (nARF) are most pronounced in sonication of medium by short ultrasonic pulses with duration on the order of microseconds. Experiments with tissue-mimicking gelatin blocks and various excised animal tissues demonstrated the possibility of measuring sub-nanometer range displacements induced by nARF acting on interfaces in materials with different compressional wave speeds. A continuous wave Doppler device was used to measure the interface particles velocity. Demonstrated possibility of using Doppler signal processing to extract the nanometer-order motion induced by the nARF could open new areas for ultrasound imaging and tissue characterization. The creation of multi-parametric images of soft tissue using nARF for obtaining maps of the compressional wave speed and shear modulus is considered.

8:30

2aBAa3. Relative contributions of various modes of acoustic radiation force to creating mechanical stress in soft tissues. Armen Sarvazyan (Artann Labs., Lambertville, NJ) and Lev Ostrovsky (Earth System Res. Lab., NOAA, 325 Broadway, Boulder, CO 80305, lev.a.ostrovsky@noaa.gov)

There are several physical effects responsible for the generation of acoustic radiation force (ARF) in soft tissue. The first effect, which is widely used in elastographic applications of ARF, is the transfer of momentum from ultrasonic wave in an attenuating medium due to absorption and scattering. Two other effects include reflection of sound wave from the boundaries between tissues differing in the acoustic impedance and second, spatial variations of the energy density of the propagating wave as a result of variations in the sound wave velocity in a medium. The latter mode of ARF can be directed both outward and toward the source of the propagating wave. In this study, relative contributions of various modes of acoustic radiation force are investigated both theoretically and experimentally. Using Green function for the acoustic potential near the boundaries between tissue structures, the expression for the motion of tissue interfaces under the action of short ultrasonic pulses is derived. Theoretical estimates of dynamics of the soft tissue response to various modes ARF induced by the microsecond range ultrasonic pulses are in agreement with experimental data. Findings of this study may serve as the basis of new approaches in soft tissue elastography.

8:45

2aBAa4. Harmonic modulation of acoustic standing wave fields for tissue engineering. Timothy E. Doyle, Blaine Johnson, Brian D. Patchett (Phys., Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Natalie C. Sullivan (Chemistry, Utah Valley Univ., Orem, UT), Dolly A. Sanjinez, Ashley Behan (Biology, Utah Valley Univ., Orem, UT), Nicole Cowan (BioTechnol., Utah Valley Univ., Orem, UT), and Audrey P. Butler (Univ. of Utah, Salt Lake City, UT)

Acoustic standing wave fields can be used to produce virtual, scaffoldless templates for tissue patterning and engineering. To date, only simple tissue patterns have been attempted and achieved such as layered microstructures. This is due in large part to the use of single-frequency standing waves transmitted from a single source, producing planar nodes and antinodes in the cell/growth medium suspension. The purpose of this study was to explore with computational models the possibility of generating complex tissue microstructures using multiple sources with multiple frequency capabilities. The models simulated multifrequency standing waves (i.e., modulated by higher harmonics), multiple acoustic sources at various angles to each other, and corresponding acoustic cavities. The simulations demonstrated that interference patterns of standing waves could be produced with a three-dimensional complexity similar to that of natural biological structures. Such patterns included alveolar structures (thin, membrane-like walls encapsulating open cavities), lobular structures (cell clusters interlaced with duct- and vascular-like channels), and radial structures (iris-like patterns). Experiments for rendering these models to actual tissue structures have also begun, including the development of multifrequency transducers, using multiple transducers to generate a square lattice of cell walls and channels, and methods to reproduce natural tissue structures with high fidelity.

9:00

2aBAa5. Bone demineralization assessment using acoustic radiation force. Max Denis (Mayo Clinic College of Medicine, 1 University Ave., Lowell, MA 01854, max_f_denis@hotmail.com), Leighton Wan, Mathew Cheong, Mostafa Fatemi, and Azra Alizad (Mayo Clinic College of Medicine, Rochester, MN)

In this work, an ultrasound-guided remote measurement technique is proposed for bone demineralization assessment. Utilizing an acoustic radiation force (ARF) beam as our excitation source and a receiving hydrophone, the mechanical properties of a bone can be noninvasively assessed. Focusing the ARF beam on the bone surface acts as point force generating vibrational waves. Coupling the bone surface and hydrophone, the ensuing radiating acoustic pressure from these vibrational waves are captured for analysis. Of particular interest, are the features best related to the bone's mechanical properties. Conducting *ex-vivo* experiments demonstrated that the velocity feature best delineates intact and demineralized bones. The typical velocity of an intact bone (3000 m/s) is higher in comparison to a 72 h demineralized bone (1600 m/s). According to the receiver operating characteristic (ROC) curve, the optimal velocity cut-off value of 3096 m/s yields 80% sensitivity and 82.61% specificity between the intact and demineralized bones. Other features, such as the spectra of the demineralized bones' acoustic response, exhibited higher attenuation for frequencies below 200 kHz in comparison to the intact bones. A time-frequency analysis demonstrated a frequency shift with demineralization. These results demonstrate the potential application of our proposed technique for monitoring bone demineralization.

9:15

2aBAa6. A generalized reconstruction framework for transient elastography. Mahdi Bayat (Physiol. and Biomedical Eng., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu), Susanta Ghosh (School of Eng., Duke Univ., Durham, NC), Azra Alizad (Physiol. and biomedical Eng., Mayo Clinic, Rochester, MN), Wilkins Aquino (Dept. of Civil and Environ. Eng., Duke Univ., Durham, NC), and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Ultrasound transient elastography has emerged as a promising imaging modality for noninvasive evaluation of tissue mechanical properties. Most of the established techniques, however, rely on a pointwise measurement of the induced shear wave speed based on a localized planar wave propagation assumption. This assumption is not always true as most tissues exhibit heterogamous characteristics. Complex interaction of different geometries, boundary conditions, and tissue composition limit the ability of these techniques such that additional spatial-temporal filtering and signal truncations are required. These signal conditionings might lead to improvement in elasticity estimation for some cases (e.g. certain geometries, material properties) but the fact that they do not stem from a comprehensive framework can lead to unrealistic mechanical properties in more complex scenarios. We present a generalized framework based on inverse solution of the elasto-dynamic equations which can simultaneously solve for material properties and geometries without requiring any prior knowledge of the wave types. The method is based on a modified error in constitutive equation (MECE) without requiring additional filtering. We present the results of standard phantom studies based on acoustic radiation force excitation and fast ultrasound tracking where MECE technique successfully estimated the elastic moduli and geometry. [Work supported by NIH grant CA174723.]

9:30

2aBAa7. Differentiation of breast lesions based on viscoelasticity response at sub-Hertz frequencies. Mahdi Bayat, Alireza Nabavizadeh, Viksit Kumar, Adriana Gregory, Azra Alizad, and Mostafa Fatemi (Biomedical and Physiol., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu)

There is an urgent need for a noninvasive but more accurate method to assess breast masses before referring the patient to biopsy. In this study, we use the slow viscoelastic creep response to evaluate breast masses at low ($<$1 Hz) frequencies. This method uses an automated ramp-and-hold compression device and a high-frame-rate ultrasound imaging to track tissue strain. The study cohort included 30 pre-biopsy patients with suspicious masses. Sequential high-frame-rate images were recorded during the ramp-and-hold process. The data were used to calculate the creep response at each point. Then, using a classic linear model, the retardation-time map was created within the field of view. The resulting retardation-time maps exhibited clear distinction of the mass margins. The results showed that benign breast lesions appear with an increase in creep retardation-time compared to surrounding breast glandular tissue, and the opposite trend was true for the malignant lesions. Statistical analysis of the viscoelasticity features revealed that a contrast feature based on the retardation-time is an accurate classifier of malignant and benign masses ($P<0.0003$). It is concluded that the retardation time is a promising biomarker for differentiation of breast masses. [Work supported by NIH Grant CA168575.]

Session 2aBAb

Biomedical Acoustics: Therapeutic Ultrasound and Microbubbles

Parag V. Chitnis, Chair

Department of Bioengineering, George Mason University, 4400 University Drive, 1G5, Fairfax, VA 22032

Contributed Papers

10:00

2aBAb1. Jet formation of contrast microbubbles in the vicinity of a vessel wall. Nima Mobadersany and Kausik Sarkar (George Washington Univ., 800 22nd St. NW, SEH 3000, Washington, DC 20052, sany@gwmail.gwu.edu)

Behaviors of microbubble contrast agent near a vessel wall under ultrasound excitation are investigated using a boundary integral method. Ultrasound in the presence of microbubbles facilitates drug delivery by streaming and jetting phenomena. The microbubbles are encapsulated by a layer of proteins or lipids to stabilize them against dissolution. The encapsulating shell of the contrast microbubble is viscoelastic modeled here by three different interfacial rheological models. While at low excitations microbubbles undergo regular oscillations, at high amplitudes excitation, they form jets toward the wall. The dynamics and the resulting shear stress on the wall are studied varying the shell rheology and other relevant parameters of the problem.

10:15

2aBAb2. An Eulerian-Lagrangian study of cloud dynamics near a wall. Jingsen Ma, Georges L. Chahine, and Chao-Tsung Hsiao (Dynaflow, Inc., 10621-J Iron Bridge Rd., Jessup, MD 20794, jingsen@dynaflow-inc.com)

The dynamics of bubble clouds is studied using an Eulerian-Lagrangian model treating the two-phase medium as a continuum and modeling the microbubbles as discrete sources and dipoles tracked in a Lagrangian fashion. These two are coupled through the local void fractions associated with the instantaneous bubble volumes and locations. Resonance of the bubble cloud, resulting in the highest pressure at the nearby rigid wall, is deduced from simulations with the same initial bubble distribution while varying the excitation frequency and amplitude. This resonance frequency deviates significantly from the classical linearized solution as the relative amplitude of driving pressure increases. It gradually drops as the excitation amplitude increases until reaching a limit value. In the high amplitude driving pressure regime, the peak pressure generated at the wall has a maximum for an optimum value of the initial bubble radius for the same cloud radius. This occurs when the ratio of maximum over initial bubble size is maximum. Too weak or too strong bubble interactions in the cloud inhibit strong collective effects. For the same initial void fraction and for different initial bubble sizes (changing the bubble number), an optimum bubble size also exists, resulting in the highest collective pressure.

10:30

2aBAb3. Optical property changes in *ex vivo* tissues exposed to high-intensity focused ultrasound. Jason L. Raymond, Eleanor Edwards, Robin O. Cleveland, and Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, 17 Parks Rd., Oxford OX1 3PJ, United Kingdom, jason.raymond@eng.ox.ac.uk)

Real-time acousto-optic (AO) sensing has been shown to non-invasively detect changes in *ex vivo* tissue optical properties during high-intensity focused ultrasound (HIFU) exposures. Baseline changes in optical properties have been previously measured as a function of thermal-dose for tissues exposed to a temperature-controlled water bath (doi:10.1088/0031-9155/59/13/3249). In this work, the wavelength-dependent optical scattering and absorption of *ex vivo* tissues exposed to HIFU were measured using an integrating sphere spectrophotometric technique employed previously. HIFU-induced spatiotemporal temperature elevations were measured using an infrared camera and used to calculate the thermal dose delivered to a localized region of tissue. We consider the impact of thermal dose, temperature elevation, and heating rate on the formation of HIFU lesions and the resulting changes in tissue optical properties. Recently, reported results (doi:10.1088/0031-9155/59/13/3249) refute the hypothesis that optical property changes in tissue are based solely on accumulated thermal dose—this claim will be explored further through the optical characterization of lesions formed in clinically relevant tissues. Results will show how wavelength-dependent optical property changes in tissues can be used to improve the AO sensing of lesion formation during HIFU therapy as an alternative to thermometry. [Work supported by the ASA F.V. Hunt Fellowship and the University of Oxford.]

10:45

2aBAb4. Tracking kidney stones during shock wave lithotripsy. Kya Shoar (Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, kya.shoar@magd.ox.ac.uk), Ben Turney (Nuffield Dept. of Clinical Medicine, Univ. of Oxford, Oxford, United Kingdom), and Robin Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Shock wave lithotripsy is a non-invasive procedure by which kidney stones are fragmented by shock waves. Real-time monitoring during shock wave lithotripsy remains a challenge more than three decades after it was introduced into clinical practice. Currently, many shock waves are delivered to the body that do not impact the stone, but do result in tissue trauma. This work presents a monitoring system to locate kidney stones, with the goal of gating shock waves not aligned with the stone, and hence, reduce renal trauma during lithotripsy. A circular array, housing twenty-two 0.5 MHz transducers that can be mounted on a clinical lithotripter, was deployed in a water tank. An algorithm, consisting of threshold detection, including automatic rejection of weak signals, and triangulation, was developed to determine the location of stones. The accuracy of the system was tested using: a spherical steel ball and two stone models made from gypsum cement. The results show that within ± 15 mm of the focus of the lithotripter, the accuracy was better than 5 mm in the lateral directions and 2 mm in the axial direction. Using off-the-shelf hardware, the algorithm can calculate stone position every 1.1 s, allowing for real-time tracking during lithotripsy. [Work supported in part by NIH through P01-DK43881.]

11:00

2aBAb5. Effects of acoustic parameters on nanodroplet vaporization. Krishna N. Kumar, Mitra Aliabouzar, and Kausik Sarkar (George Washington Univ., 800 22nd St. NW, SEH 3000, Washington, DC 20052, krishnasst@gmail.com)

Phase shift nanodroplets offer a number of advantages over ordinary microbubbles due to their enhanced stability and smaller size distribution. These nanodroplets undergo a phase transition from liquid to highly echogenic gaseous state when activated by sufficient acoustic energy through a process termed acoustic droplet vaporization (ADV). In this study, we synthesized lipid-coated perfluoropentane (PFP) filled nanodroplets via sonication

method. We investigated the ADV threshold of these nanodroplets as a function of acoustic parameters such as excitation pressure, frequency, pulse length, and pulse repetition period (PRP). Our results indicate that ADV threshold varies significantly with PRP; while at PRP of 10 ms, the ADV threshold was found to be 3.6 MPa (pk-pk), for PRP of 1 ms, 100 μs, and 500 μs, ADV was not observed even at 10 MPa. At ADV, fundamental and odd harmonics were found to be significantly higher than the background noise. The acoustic response of ordinary perfluorobutane (PFB) filled microbubbles with the same lipid composition is compared to that of PFP nanodroplets when both were excited with the same excitation pressures (450 kPa to 10 MPa). Above ADV threshold both showed a similar response.

11:15

2aBAb6. Effects of ultrasound in presence of microbubbles on cartilage tissue regeneration in three-dimensional printed scaffolds. Mitra Aliabouzar, Lijie G. Zhang, and Kausik Sarkar (George Washington Univ., 800 22nd St. NW, SEH 3000, Washington, DC 20052, mitra.aalee@gmail.com)

Gas-filled microbubbles encapsulated with lipids and other surfactants are highly responsive to ultrasound, which has led to their effective role as ultrasound contrast agents (UCA). In this study, for the first time, we used lipid-coated microbubbles (MB) prepared in-house in order to better harness the beneficial effects of ultrasound stimulation on proliferation and chondrogenic differentiation of human mesenchymal stem cells (MSCs) within a novel 3D printed poly (ethylene glycol) diacrylate (PEG-DA) hydrogel scaffolds. A significant increase in cell number (p <0.001) was observed with low intensity pulsed ultrasound (LIPUS) treatment in the presence of 0.5 % (v/v) MB after 1, 3, and 5 days of culture. MSC proliferation enhanced up to 40% after 5 days of culture in the presence of MB and LIPUS while this value was only 18% when excited with LIPUS alone. We investigated the effects of acoustic parameters such as excitation intensity, frequency, and pulse repetition period on MSC proliferation rate. Our 3-week chondrogenic differentiation results demonstrated that combining LIPUS with MB significantly enhanced both Glycosaminoglycan (GAG) and type II collagen production. Therefore, integrating LIPUS and MB appears to be a promising strategy for enhanced MSC growth and chondrogenic differentiation.

11:30

2aBAb7. Microbubble response to dual frequency excitation for broadband contrast imaging. Christina Keravnou (Dept. of Bioeng., Univ. of Washington, Benjamin Hall University of Washington, 616 NE Northlake PL., Seattle, WA 98105, ckerav@uw.edu), Chrysovalantis Papantonis (Dept. of Mech. and Manufacturing Eng., Univ. of Cyprus, Nicosia, Cyprus), and Michalakis Averkiou (Dept. of Bioeng., Univ. of Washington, Seattle, WA)

Contrast imaging relies on pulsing schemes like pulse inversion (PI), power-modulated pulse inversion (PMPI), and power modulation (PM) to isolate certain harmonic components. These multi-pulse schemes normally employ single frequency pulses (SFP). In this work, we propose dual frequency pulses (DFP) coupled with pulsing schemes (PI, PMPI, and PM) for microbubble imaging. The proposed DFP contain an amount b_1 of a fundamental frequency f and an amount b_2 of its phase-shifted (φ) second harmonic $2f$. Microbubbles were excited with a variety of DFP combinations (b_1, $b_{2,}$ φ) and their nonlinear response was compared to that from conventional SFP. All pulsing schemes benefit in terms of overall signal from the use of DFP by ~4–6 dB. PI benefits the most when coupled with DFP as it extracts an additional amount of nonlinear fundamental signal (~25 dB) something not feasible with SFP. It was observed that microbubble nonlinear response changes when the relative amount of f and $2f$ in DFP change, allowing the design of specific pulses for different microbubbles. In conclusion, since the use of DFP with pulsing schemes extracts harmonics more efficiently than SFP it has the potential to improve contrast imaging in terms of bubble sensitivity and image resolution.

11:45

2aBAb8. Lytic efficacy of tissue plasminogen activator and ultrasound in porcine clots doped with barium sulfate *in vitro*. Shenwen Huang, Himanshu Shekhar, and Christy Holland (Univ. of Cincinnati, 231 Albert Sabin Way, Mail Location: 0586, Cincinnati, OH 45229, huangsw@mail.uc.edu)

Swine and porcine tissue are employed routinely for preclinical models of ischemic stroke. In this study, we examined the lytic susceptibility of porcine whole-blood clots, prepared with and without barium sulfate, a radiopaque x-ray contrast agent. The degree of lytic efficacy in the two types of clots was compared for recombinant tissue plasminogen activator (rt-PA) concentrations ranging from 0 to 100 μg/mL. Subsequently, a *in vitro* flow model and time-lapse thrombolysis microscopy system was used to evaluate lysis in clots with and without barium sulfate in response to rt-PA and 120-kHz intermittent ultrasound exposure. The degree of lysis observed with both types of clots was similar for rt-PA concentrations up to 15.75 μg/mL. However, clots doped with barium sulfate demonstrated significantly lower lysis at higher rt-PA concentrations. Similarly, results obtained using the *in vitro* flow model showed that both types of clots underwent comparable lysis when treated with 15.75 μg/mL rt-PA. Further, using Definity® and 120-kHz ultrasound as an adjuvant to rt-PA did not enhance lysis in either porcine clot model compared to rt-PA alone. These results highlight the considerable differences in the degree of clot lysis between porcine clots and previously reported studies that used human clots.

Session 2aEA

Engineering Acoustics: General Topics in Engineering Acoustics I

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Chair's Introduction—8:00

Contributed Papers

8:05

2aEA1. Characterization of porous material and a sound suppression system using a shock tube. Erica K. Good, Robert Taylor, James Luskin, and Zachary Conaway (College of Eng., The Catholic Univ. of America, 655 Michigan Ave. NE, Apt. P502, Washington, DC 20017, 45taylor@cua.edu)

This work describes the design and construction of an experimental apparatus for testing shock waves and a sound suppression system for student learning. The shock tube will be used to characterize both porous media and the suppression system response to a non-linear shockwave. The effects of a plane shock wave will be used to determine acoustic properties, such as transmission, reflection, and absorption coefficients, of rigid frame porous materials. The suppression system is comprised of 20 conical chambers that cause a loss of energy in the shock wave by the expansion and acceleration of the flow. The sound pressure level from the shock is in excess of 140 dB re 20 μPa. Material properties and the performance of the designed suppression system will be presented.

8:20

2aEA2. A method to suppress harmonics in standing-wave thermoacoustic engines. A. H. Ibrahim, M. Emam (American Univ. in Cairo, New Cairo 11835, Egypt, abdelmaged@aucegypt.edu), A. M. Fouad (Mech. Power Dept., Cairo Univ., Giza, Giza, Egypt), and Ehab Abdel-Rahman (American Univ. in Cairo, New Cairo, Egypt)

Harmonic generation in thermoacoustic engines is regarded as a non-linear loss mechanism that extracts acoustic power from the fundamental wave into harmonics. It is shown that the use of specifically designed insert that limits the gas flow area over a very-limited part of the resonator can suppress this non-linear loss mechanism causing a significant increase in the generated acoustic power in the fundamental mode. In this experimental work, a standing-wave thermoacoustic engine is built and operated without inserts and with inserts of different shapes, porosities, and thicknesses. The self-generated dynamic pressure waves are captured under different operating conditions and then are decomposed into a fundamental component and harmonics. Results for different inserts are presented and discussed. All inserts caused lower harmonic content with respect to the no-insert case but inserts of low gas flow area cause severe flow blockage and thus a severe reduction in the produced acoustic powers. Results analyze the relationships between the generated acoustic power in the fundamental mode and the amplitudes of the dynamic pressures of the fundamental, first harmonic, and second harmonics. The blockage effects caused by the insert are discussed.

8:35

2aEA3. Efficient dispersion analysis of guided waves in laminated composites and substrates. Ali Vaziri and Murthy N. Guddati (Civil Eng., North Carolina State Univ., 208 Mann Hall, 2501 Stinson Dr., Campus Box 7908, Raleigh, NC 27695-7908, avaziri@ncsu.edu)

Nondestructive testing of laminated composites and substrates involve matching observed and predicted dispersion characteristics of guided waves. These characteristics are quantified through dispersion curves (phase velocity versus frequency) and need to be computed for a large number of estimated structure and material property combinations. Given its central nature, we propose an efficient approach for computing the dispersion curves, leading to an order of magnitude savings in the computational cost. Our approach is based on conventionally used finite element semi-discretization through the depth, but with one significant modification: by using a specially designed set of complex-valued finite element lengths through the depth, we show that the dispersion curves can be obtained with a handful of elements per layer as opposed to larger number of traditional finite elements, resulting in large reduction in computational effort. In this talk, we present the formulation of the proposed complex-length finite element method and illustrate its efficiency through modeling wave dispersion in laminated composites and substrates. Finally, we introduce an inversion procedure developed around this method and demonstrate its effectiveness in characterizing plate structures using synthetic as well as real nondestructive testing data.

8:50

2aEA4. Reverberation characterization inside an anechoic test chamber at the weapon sonar test facility at Naval Undersea Warfare Center Keyport Division. Grant C. Eastland and William C. Buck (Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

The Weapon Sonar Test Facility (WSTF) at NUWC Keyport, WA, is a 34500 gallon pressure tank currently used for test and evaluation of torpedo sonar arrays. The tank has many more possible uses and complete characterization of the testing environment needs to be performed. One of the methods of characterization being used is the determination of the reverberation time of the anechoic chamber. Applying Sabine-Franklin-Jaeger theory of reverberant rooms, an experimentally determined reverberation time T_{60} for a "live room" can be used to provide an upper bound for the reverberation time in the chamber. Utilizing the Eyring theory of "dead" rooms will provide a better determination of T_{60} for the anechoic chamber, hence provide a lower bound characterization. The experimental method involves determining the spatially averaged acoustic energy of an initial tone and the corresponding reflection from the chamber wall seen in a recorded signal. These values determined from the root mean square signal voltage determine the wave decay time, τ. The decay time is directly related to the reverberation time through $T_{60} = 6\tau\times\log_e 10$. The reverberation was determined to be within an acceptable tolerance for testing and evaluation.

9:05

2aEA5. A comparison of different methods for calculating complex acoustic intensity. Eric B. Whiting, Kent L. Gee, Scott D. Sommerfeldt, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, ericbenwhiting@gmail.com)

The Phase and Amplitude Gradient Estimation (PAGE) method of calculating acoustic intensity from multiple pressure measurements [Thomas *et al.*, JASA **137**, 3366–3376 (2015)] has been used successfully to calculate active intensity in the laboratory and in-field experiments. The primary result is that the PAGE method increases the bandwidth over which accurate vector intensity estimates can be obtained. Further work has investigated the application of the PAGE method to calculate other energy-based quantities, including the reactive intensity. Simulated pressure and particle velocity fields have been calculated for a single monopole, two out-of-phase monopoles, and, via Rayleigh integration, a baffled circular piston. The analytical intensity fields are then compared to the PAGE and the traditional p-p methods for obtaining both the active and reactive intensity, both in the near and far fields. [Work supported by NSF.]

9:20

2aEA6. Study of virtual instrument technology applied in sound field test. Yongchao Yao, Xiaodong Ju, Junqiang Lu, and Baiyong Men (China Univ. of Petroleum (Beijing), No.18, Fuxue Rd., Changping District, Beijing, Beijing 102249, China, Wardruna2013@163.com)

As a widely used technology in the field of test and measurement, virtual instrument (VI) combines high-performance modular hardware with flexible software to perform a variety of experimental tasks. In order to perform the sound field test in laboratory, we build an specialized platform of VI, whose hardware system mainly consists of multi-channel data acquisition boards, independence analyzer, mechanical positioning device, and industrial personal computer equipped with PXI, GPIB, and Ethernet bus interface. Also, a graphical VI programming language (LabVIEW) is adopted to develop the corresponding software system. With this VI platform, we measure the impedance property of transducers and the sound field of a phased array composed of them. The results show that the transducers are positioned correctly by the mechanical positioning device, the excitation time of every transducer of the phased array is controlled accurately, and finally a set of high-quality waveforms are acquired.

9:35

2aEA7. Loudspeaker system. Gordon Pisani, Nico Bartolomeo, Joseph Mascolo, Gabrielle Wolff, Alexander Laprade, Ethan Gorczynski, and Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 17pisani@cua.edu)

This presentation will demonstrate the design, construction, testing, and final performance of a student designed loudspeaker system. The purpose of this project was to create a 2.1 channel, high quality sound system that produces a bandwidth just short of the full range of human hearing, approximately 20–20000 Hz. This system consists of 2 two-way speakers utilizing a woofer and tweeter, with one separate subwoofer. The function of this particular sound system is to reproduce sound faithfully in a 350–400 square foot room. The performance characteristics of this system will be presented through frequency response and linearity testing results. These tests were used to calculate and implement modifications made to the cabinet size and shape while adhering to a budget of $1200. The analysis of every performance will also reveal data for different electrical component values in designing the Linkwitz-Riley crossovers.

9:50

2aEA8. Jet pump oscillating-flow behavior at different Womersley numbers. Abdelrahman Nassif, Ahmed I. Abd El-Rahman, and Ehab Abdel-Rahman (Phys., American Univ. in Cairo, AUC Ave., New Cairo 11835, Egypt, ahmedibrahim@aucegypt.edu)

Few numerical works simulate the *laminar*-flow behavior within the diverging passages of typical thermoacoustic jet pumps. The associated boundary-layer separation is mostly delayed by maintaining a slowly increasing flow passage and thereby the corresponding pressure drop and consumed acoustic energy are noticeably reduced. Of equal importance is the effect of the *transition to turbulence* on the boundary-layer separation and the resulting flow-pattern. To our knowledge, no simulation has been developed that numerically predict and capture the conditionally turbulent flow regime within typical jet pumps. Here, a new finite-volume model is reported that employs the non-linear CFD solver of ANSYS FLUENT. The k-kl-ω transition model is considered and its parameters are particularly adjusted for present application. A sinusoidal pressure oscillation is enforced at one end, while the far-field approximation is considered at the other end to model the non-reflecting boundary condition. An axisymmetric model along with careful meshing are applied. The transient simulation run proceeds till stationary flow behavior is obtained. Both the flow streamlines and vorticity field are plotted for three different acoustic frequencies, Womersley numbers of which correspond to 5, 15, and 30. The numerical results capture the onset of flow transition into turbulence and characterize the flow patterns at different frequencies. The results also illustrate the influence of the developed turbulent flow-patterns on the boundary-layer separation.

10:05–10:20 Break

10:20

2aEA9. Classification and optimization of beam responses synthesized from non-uniformly spaced linear arrays. Chrisna Nguon (Univ. of Massachusetts Lowell, Dracut, MA), Nicholas Misiunas, Jenny Au, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, Nicholas_Misiunas@student.uml.edu)

A probability distribution for the element positions on a linear array is derived so as to match a target beam pattern that approximates a Dolph-Chebychev array. The random beam responses generated from such a configuration exhibit a mean profile in observation angle that is invariant with element size N but the variance decays as 1/N. The beam responses are classified considering the N-1 inter-element distances as features that control the deviation from the mean beam profile and the variance of the class. The inter-element distance constraints are incorporated in a computational model of the firefly algorithm (FA) designed to optimize the location of the array elements that match the target beam pattern. The FA consists of a set of fireflies, representing arrays, which move through the solution space with directed movement based on each fireflies' fitness and random movement to ensure adequate exploration and avoid local minima. The fitness function is a combination of a target beam-width, side-lobe level, and bounds on the position dependent mean inter-element distances. The variance of the resulting beams are compared with the result from the unconstrained random placement of elements as a function of the element size and side-lobe parameters.

10:35

2aEA10. Analysis of the error sources of the two-microphone transfer function method for measuring absorption coefficient in the free field using numerical modeling. Hubert S. Hall, Joseph F. Vignola, John A. Judge, and Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 61hall@cua.edu)

Extension of the two-microphone transfer function absorption coefficient measurement technique from impedance tubes to the free field introduces several error sources. The three-dimensional nature of the sound field necessitates consideration of factors that are not relevant in an enclosed tube below the cutoff frequency. The impedance tube technique has been modified to account for non-planar wave propagation due to an acoustic point source. The sound field contamination from sample edge diffraction has generally restricted use of the technique to frequencies such that wavelengths are small relative to sample dimensions. This requires the use of very large test panels for low frequencies. Numerical models of the two-microphone free field technique have been created to quantify these effects. Each effect was isolated to better understand its independent impact on the accuracy of the technique. Finally, a series of experimental tests were conducted to validate the numerical modeling results.

10:50

2aEA11. Optimizing an underwater transmit array of projectors with large variations. Eunghwy Noh, Wonjong Chun, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ohm@yonsei.ac.kr), and Youngsoo Seo (Agency for Defense Development, Changwon, Gyeongsangnam-do, South Korea)

Uniformity of constituent projector elements is a prerequisite for building a high-performance underwater transmit array. Unless the element-to-element variations in transmitting voltage response (TVR) and directivity fall within specified limits, the resulting acoustic performance of the array will suffer significantly in terms of its sound output and directivity. Here, we consider a case in which projector elements exhibit a large scatter in TVR, say, over 6 dB, and present a design guide to still get the most of the array. The focus of the study is to explore how the relative placement of projector elements affects their mutual interaction and ultimately leads to the overall characteristics of the array. Finite element computations are carried out to derive an optimized design and the design guide. [This work has been supported by the Low Observable Technology Research Center program of Defense Acquisition Program Administration and Agency for Defense Development.]

TUESDAY MORNING, 24 MAY 2016 — SALON I, 10:25 A.M. TO 11:55 A.M.

Session 2aED

Education in Acoustics: Education in Acoustics General Topics

Jon P. Johnson, Cochair
Physics, Brigham Young University - Idaho, 525 S Center St., Rexburg, ID 83460-0520

Ryan Nielson, Cochair
Brigham Young Univ. - Idaho Physics, 525 S. Center St., Rexburg, ID 83460-0520

Contributed Papers

10:25

2aED1. Cultivating successful undergraduate research by cognitive apprenticeship: A case study. Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu)

A common challenge in academic research is finding a meaningful and productive way to involve undergraduate students in substantive research activities. The Department of Engineering at East Carolina University offers a general Engineering undergraduate degree, and matriculated its first small group of graduate students into a Master's program in Biomedical Engineering in Fall of 2014. With few master's students and no doctoral program, our undergraduate students are often given the opportunity to play a role in research that is more typically expected of a traditional graduate research assistant. This fact presents both great opportunity and unique challenges. This work presents the framework for a case study in cognitive apprenticeship that will take place over the next four years. The group of students includes all class levels, from freshmen to seniors. In addition, implementation of a hive or collaborative approach for introducing undergraduate students to advanced technical literature is described.

10:40

2aED2. An open-access interactive textbook for teaching room acoustics modeling. Lauri Savioja (Dept. of Comput. Sci., Aalto Univ., PO Box 15500, Aalto FI-00076, Finland, Lauri.Savioja@aalto.fi)

There are several different techniques for modeling room acoustics. Interactive visualizations can be used to ease learning the main principles and algorithms for that. This presentation will demonstrate a new interactive open-access textbook aiming to support learning the basics of room acoustics modeling techniques. The visualizations illustrate how different algorithms work in simple 2-D geometries, and what are the underlying assumptions in each of them. Interactivity here means that the students have control over both various parameters that affect the modeling performance and accuracy, and the geometries. They can study how the modeling results change in response to such changes. The main emphasis here is on geometrical room acoustics, and the techniques based on that: image-source model, ray-tracing, and acoustic radiance transfer, but the longer term aim is to cover all the main room acoustic modeling principles and techniques.

10:55

2aED3. The sound of STEAM (science, technology, engineering, arts, and math): Acoustics as the bridge between arts and STEM (science, technology, engineering, and math). Caleb Goates, Jenny Whiting (Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com), Mark Berardi (Michigan State Univ., East Lansing, MI), Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

This paper describes the development and presentation of a Science, Technology, Engineering, Arts, and Math (STEAM) workshop for elementary school teachers designed to provide ideas and tools for using acoustics in the classroom. The abundant hands-on activities and concepts in acoustics naturally link science and music in an intuitive way that can assist teachers in moving forward on the STEAM initiative. Our workshop gave teachers an introduction to acoustics principles and demonstrations that can be used to tie STEAM techniques in with Utah State Education Core standards. These hands-on demonstrations and real-world applications provide an avenue to engage students and support learning outcomes. Feedback indicated that the participants learned from and enjoyed the initial implementation of this workshop, though many elementary school teachers did not immediately see how they could integrate it into their curriculum. While additional efforts might be made to better focus the training workshop for the K-6 level, curriculum developers need to appreciate how acoustics could be used more broadly at the elementary school level if the emphasis changes from STEM to STEAM.

11:10

2aED4. Loudspeaker design and analysis. Joseph Mascolo (Mech. Eng., The Catholic Univ. of America, 38 Canfield Rd., Morristown, N J 07960, 30mascolo@cua.edu), Nico Bartolomeo, Alexander Laprade, Ethan Gorczynski, Gabrielle Wolff, Gordon Pisani, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This presentation will demonstrate the planning, construction, testing, and final performance of a student designed loudspeaker system. The purpose of this project was to create a 2.1 channel, high quality sound system that produces a bandwidth just short of the full range of human hearing, which is then cut off and sent to the appropriate speakers. This system consists of 2 two-way speakers utilizing a woofer and tweeter, arranged in unison with one sole subwoofer. The function of this particular sound system is to reproduce sound faithfully in a 350–400 square foot room, yet still be lightweight and maneuverable. The performance characteristics of this system will be presented through frequency response and linearity testing results. These tests were used to help warrant modifications made to the cabinet size and shape while adhering to a strict budget of $1,200. The analysis of every performance will also reveal data for different electrical component values in designing the Linkwitz-Riley crossovers. The aim of this presentation is to highlight the various key factors that are important in loudspeaker design.

11:25

2aED5. High frequency sound speed measurements in liquids and solids using a smartphone. Blake T. Sturtevant and Dipen N. Sinha (Mater. Phys. and Applications, Los Alamos National Lab., MPA-11, MS D429, Los Alamos, NM 87545, bsturtev@lanl.gov)

Much has been written in recent years on the use of smartphones and other personal electronic devices (PED) in the high school and college teaching laboratories. Due to the bandwidth limitations ($<$ 20 kHz) of these devices, experiments proposed to date have focused almost exclusively on measurements of phenomena in air and in the audible range. This talk will focus on a simple way to extend the use of these devices to measure sound speed in liquid and solids which have significantly higher sound speeds than air. High frequency acoustic standing waves in these media can be generated using standard teaching laboratory instrumentation (e.g., signal generators providing white noise, chirp signals, CW signals with manually adjusted frequency, etc.) and two opposing piezoelectric disks. By down-converting the high frequencies with an inexpensive diode mixer and measuring the difference frequency spectrum with a PED, it is possible to determine multiple successive standing wave frequencies in the medium. The measured frequencies, along with the distance between the disks, can then be used to determine sound speed accurately. Simple and inexpensive, the proposed system provides a learning experience traditionally inaccessible without research grade RF measurement instrumentation.

11:40

2aED6. A new acoustics research group and course at Brigham Young University—Idaho. Jon P. Johnson (Phys., Brigham Young Univ. - Idaho, 525 S Ctr. St., Rexburg, ID 83460-0520, johnsonjo@byui.edu), Kent L. Gee (Phys., Brigham Young Univ., Provo, UT), and Ryan Nielson (Phys., Brigham Young Univ. - Idaho, Rexburg, ID)

This paper details the beginning of a collaboration between the Acoustics Research Group at Brigham Young University in Provo, UT, and a new research group at BYU-Idaho in Rexburg, ID. BYU-Idaho is a four-year university without a graduate program. Its physics program awards 10 to 15 BS degrees in physics each year. Historically, only a 100-level acoustics course has been offered by the physics department, but in conjunction with forming this new research group we are hoping to construct an upper-division acoustics course assuming a background in differential equations, computational physics, and Fourier analysis. An overview of the collaboration, which is still in its infancy, is discussed.

TUESDAY MORNING, 24 MAY 2016 — SALON B/C, 8:00 A.M. TO 10:50 A.M.

Session 2aMU

Musical Acoustics: Voice Registration in Amplified and Unamplified Singing

Ingo R. Titze, Chair
National Center for Voice and Speech, 136 South Main Street, Suite 320, Salt Lake City, UT 84101-3306

Chair's Introduction—8:00

Invited Papers

8:05

2aMU1. The affect of audio enhancement on vocal timbre. Matthew Edwards (Voice/Musical Theatre, Shenandoah Univ., 129 Morning Glory Dr., Winchester, VA 22602, medwards09@su.edu)

In acoustic singing, performers learn to finely tune their resonance in order to control the spectral output of their voices. However, in amplified singing styles, microphone frequency response, equalization, and other editing tools can affect our perception of resonance. These effects can have positive and negative impacts on the spectral output of a singer's voice that teachers must be aware of when training Contemporary Commercial Music (CCM) artists. This presentation will include information on how audio equipment can alter the timbre of the voice, affect our perception of registration, and alter the singing power ratio.

8:35

2aMU2. Nonlinear dynamics helps explain how vowel influences register stability. Lynn M. Maxfield (National Ctr. for Voice and Speech, Univ. of Utah, 136 S Main St., Ste #320, Salt Lake City, UT 84101, lynn.maxfield@utah.edu)

Highly trained singers know what a registration shift feels like, where in their ranges it is likely to occur, and how to make small adjustments to make it less abrupt. They also know that the location of register changes differs depending on vowel choice and the style in which they are performing. Changing registers while singing can result in a significant timbral shift and, if approached unexpectedly, an abrupt jump in f_o. If the source and filter interact in a purely linear fashion, f_o should not be influenced by the shape of the vocal tract (vowel). This paper will demonstrate how nonlinear source-filter coupling may explain the strong relationship between vowel and registration. Eight volunteers performed f_o glides while altering the dimensions of their vocal tracts, predictably changing the formant frequencies. It was hypothesized that if the source and filter operated as a purely linear system, f_o stability should not be perturbed by formant/harmonic crossings in a linear system. Acoustic analysis revealed, however, that 85% of f_o instabilities were aligned with a crossing of one of the first four harmonics with the first three formants, indicating that source-filter coupling likely occurred.

9:05

2aMU3. Why fry? An exploration of the lowest vocal register in amplified and unamplified singing. John P. Nix (Music, Univ. of Texas San Antonio, One UTSA Circle, San Antonio, TX 78249, john.nix@utsa.edu)

Vocal fry is the lowest register in human vocal production. It can be defined on a psycho-acoustical basis as vocalization at fundamental frequencies below the perceptual threshold for discrete events, which occurs at approximately 70 Hz. Vocal fry has become more commonplace in conversational speech and amplified singing styles such as popular and country, but it is typically unused in non-amplified accompanied performances of most Western Classical music. The author's presentation will include the results of a survey of listener preferences of performances of popular and country performances with and without vocal fry, and the results of an experiment to examine what acoustical information is being transmitted to listeners during the fry portions of performances.

9:35

2aMU4. Vocal registers and comparative frequency analysis for multiple sung genres. Lisa Popeil (Voiceworks, 14162 Valley Vista Blvd., Sherman Oaks, CA 91423, lisapopeil@mac.com)

Diverse vocal genres utilize different registration and resonance strategies. This presentation will analyze and compare sound samples from numerous sung genres including classical (male versus female), pop, R&B, rock, country and various musical theater belting substyles (heavy belt, nasal belt, ringy belt, brassy belt, and speech-like belt). In addition, using acoustic analysis, these styles will be demonstrated by a single subject to show how registration and resonance choices can be made to clarify, identify, and codify the conventions of that genre.

10:05–10:20 Break

Contributed Papers

10:20

2aMU5. Listener ratings of singer expressivity in musical performance. Mackenzie L. Parrott and John Nix (Music Dept., Univ. of Texas at San Antonio, 1 UTSA Circle, San Antonio, TX 78249, mackenzie.lanae@gmail.com)

Vocal fry has gained a lot of attention in recent years from speech language pathologists, linguists, singers, researchers, and the general public alike. Vocal fry register uses low airflow through shortened vocal folds, causing an aperiodic pattern in vocal fold oscillation. This vocal mannerism is becoming increasingly common in American speech, fueling discussion about the implications and perception of its use. As it has become more prevalent, fry has naturally found its place in many commercial American song styles as well. Many singers are implementing fry as a stylistic device at the onset or offset of a sung tone. The objective of this study is to analyze whether listeners' ratings of a singer's expressivity in musical samples in two contemporary commercial styles (pop and country) are affected by the presence of vocal fry. We hope that this study will shed some insight into the prevalence of this particular vocalism in popular music, and to see if there is a difference in listener ratings according to the singer's gender.

10:35

2aMU6. Toward mapping voice registers in the physiological domain: A computational study. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Although there have been many studies on the acoustics and vibration of voice registers, the underlying physiological mechanisms of different registers still remain unclear. This study represents a preliminary effort toward identification of regions in the physiological domain that would lead to different voice registers, using a computational phonation model. The physiological parameters under consideration include vocal fold geometry, stiffness, resting glottal angle, subglottal pressure, and vocal tract shapes. Three registers are considered, including vocal fry, chest, and falsetto. Voice registers are identified based on known or reported acoustic features, including F0, closed quotient, and harmonic structures, based on which the physiological conditions corresponding to each voice register are determined. Preliminary perceptual experiment will be performed to confirm register identification. [Work supported by NIH.]

Session 2aNSa

Noise, Signal Processing in Acoustics, Architectural Acoustics, and ASA Committee on Standards: Noise Measurements with Mobile Apps

Benjamin Faber, Chair
Faber Acoustical, LLC, 277 S 2035 W, Lehi, UT 84043

Chair's Introduction—8:00

Invited Papers

8:05

2aNSa1. Evaluation of smartphone sound measurement applications using external microphones—A follow-up study. Chucri A. Kardous, Peter B. Shaw, and William J. Murphy (National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Cincinnati, OH 45226, ckardous@cdc.gov)

The National Institute for Occupational Safety and Health (NIOSH) conducted a follow-up study to examine the accuracy of smartphone sound measurement applications (*apps*) using external, calibrated, microphones. In the initial study, we examined 192 *apps* on the Apple (iOS) and Google (Android) platforms. Overall, 10 iOS *apps* met our selection criteria for occupational noise exposure measurements, and of those, only 4 iOS *apps* (SoundMeter, SPLnFFT, SPL Pro, and NoiSee) were within ±2 dB(A) of a reference microphone. For this study, we selected the same 4 iOS *apps* and examined their accuracy and performance using the MicW i436 and the Dayton Audio iMM-6 external microphones. The MicW i436 microphone is marketed as compliant with IEC-61672 class 2 sound level meter standard. Testing was conducted in a reverberant chamber using pink noise over seven nominal sound levels from 65 to 95 dB. The results showed an even closer agreement than the earlier study, with mean differences within ±1 dB(A) of the reference microphone. This study suggests that the use of external microphones, and the ability to calibrate the microphones and *apps*, improves the overall accuracy and precision of measurements, and removes many of the constraints and limitations associated with the built-in smartphones' microphones.

8:25

2aNSa2. Investigating the feasibility of using mobile devices for remote noise monitoring and data acquisition. Mike Dickerson (MD Acoust., 4960 S. Gilbert Rd., Ste 1-461, Chandler, AZ 85249, mike@mdacoustics.com)

The feasibility of using the Faber Acoustical SoundMeter Pro app for mobile devices for remote noise monitoring and data acquisition was evaluated and tested. A data file acquisition protocol was created, a remote database storage system was developed, and a reporting system was prepared to test the feasibility. The data and results of this assessment are presented and discussed.

8:45

2aNSa3. Assessment of noise exposures for pre-term infants during air transport to neonatal intensive care units using iPhone sound meter apps. William W. Clark (Audiol. Comm Sci., Washington Univ. School of Med, 660 S. Euclid Ave., Box 8042, St. Louis, MO 63110, clarkw@wustl.edu) and Scott Saunders (Pediatrics/Newborn Medicine, Washington Univ. School of Med., St. Louis, MO)

A significant number of infants born prematurely or with life-threatening conditions in local hospitals require transport to a regional tertiary care center. St. Louis Children's Hospital's (SLCH) neonatal intensive care unit (NICU) serves southern Missouri and Illinois, and 3000 pre-term neonates are transported to the hospital annually, most by helicopter or fixed wing aircraft. This initial study evaluates the accuracy and efficacy of using an iPhone app (SoundMeterPro,Faber Acoustics) for routine collection of infants' noise exposures inside the isolette during air transport. The app was downloaded onto iPhones (4,6S) calibrated in a sound field using a Larson-Davis type I sound level meter (831). The meter and the iPhones were placed inside an unoccupied isolette and recorded in-flight noise levels during the outbound portion of trips from the NICU to the local hospital. Sound levels were high (85–90 dBA) and higher during take-off and landings or when the isolette lid was opened, and level and OBN measures obtained with the SoundMeterPro app were similar to those from the type 1 SLM. An additional advantage of the iPhone SLM platform for routine use in aircraft is that it does not require FAA certification for each aircraft, if used in the airplane mode.

9:05

2aNSa4. 2015 Jacksonville Executive at Craig Airport noise study. Herbert R. Matthews (Aviation, Jacksonville Univ., 1806 Pleasant Point Ln., Jacksonville, GA 32225, bmatthe2@jacksonville.edu) and Ross Stephenson (Aviation, Jacksonville Univ., Jacksonville, FL)

This research will update outdated information as identified in the original 2006 Federal Aviation Regulation (FAR) Part 150 Study of Craig Municipal Airport. The original study conducted an evaluation of the Airport's existing noise conditions from 2004 to 2005 to determine if current voluntary operational procedures were achieving their desired effect, and identified other opportunities to reduce aircraft-related noise impacts on the communities surrounding the airport. This study will examine and revalidate the findings of the original 2006 study. It is expected that the noise level contours, as depicted within the original study, have reduced in size since 2006. This is due to the utilization of more technically advanced aircraft engines on today's modern business jets. Airport noise magnitude, frequency, and duration will be measured during daily operations. Sound exposure level (SEL) metric used by the Federal Aviation Administrations will be used to measure single event noise exposure levels along with time weighted cumulative noise metrics. The findings of the study may provide the opportunity for the Jacksonville Airport Authority (JAA), the aviation industry, affected political jurisdictions and airport neighbors to work together in the evaluation of potential noise reduction and land use control measures.

9:25

2aNSa5. Real-time sound measurements of exercise classes with mobile app demonstrate excessive noise exposure. Sumi Sinha, Elliott D. Kozin, Matthew R. Naunheim, Samuel R. Barber, Kevin Wong (Otolaryngol., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114, sumi_sinha@meei.harvard.edu), Leanna W. Katz (Spaulding Rehabilitation Hospital, Boston, MA), Ishmael J. Stefanov-Wagner, and Aaron K. Remenschneider (Otolaryngol., Massachusetts Eye and Ear Infirmary, Boston, MA)

Noise induced hearing loss is a major contributor to observed hearing impairment in the general population. Advanced mobile technology now allows the opportunity to directly measure noise exposure profiles on an individual basis. Herein, we conduct a pilot study to determine noise exposure in popular gym-based exercise cycling classes where music and equipment generate high intensity sound. A calibrated iOS mobile app (SoundMeter, Faber Acoustical) coupled to a microphone was used to measure noise in a random sampling of similar indoor cycling classes ($n=7$). The average length of exposure was 52.33 ± 3.81 min. Maximum sound levels recorded across all classes were 125.96 dB and averaged 119.6 ± 3.5 dBA. By NIOSH standards, 31.5 ± 14.7 min were spent over 100 dBA, corresponding to a class daily exposure dose of 191% $\pm$ 1.2%. The 8-h projected dose was 1781% $\pm$ 11.9%. Preliminary data suggest that certain exercise classes may expose participants to excessive, potentially dangerous sound levels. The use of readily accessible mobile technology can assess noise exposure risk from the consumers' perspective, allowing a new level of independent self-monitoring of noise and empowering individuals to become active participants in their hearing health.

2a TUE. AM

Contributed Papers

9:45

2aNSa6. Noise illustrated. Michael Ermann (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu)

Principles and research findings on the subject of noise control are presented in the graphic language of architecture in the hope that not only architects, but others will benefit from the translation. (1) Mass, airtightness, and structural discontinuity are illustrated with baseline/better comparisons; (2) transmission loss data are culled and translated from tables to graphs for easy contrasts, (3) families of assemblies are illustrated and ranked by STC and IIC values, (4) the frequency domain of impact noise is explored graphically, (5) barrier design principles are demonstrated, (6) AHU fan noise and turbulence attenuation strategies are presented visually, and (7) common in-the-field construction mistakes are drawn to illustrate flanking paths, vibration isolation short-circuiting, and barrier construction acoustic bridging. Rules-of-thumb and nomographs are offered for early-design best-practices in building site selection, duct distances, ducted air velocities, window selection, and space planning. The illustrations presented come from over 250 drawings in the book, *Architectural Acoustics Illustrated* (Wiley, 2015), half of which is dedicated to noise control.

10:00

2aNSa7. A comparison of readily available sound level apps with respect to field performance and applications in architectural education. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and Emily McGlohn (School of Architecture, MS State Univ., MS State, MS)

This investigation was driven first by a suggestion that apps be used by residents, sound creators, and police to evaluate sound sources in regard to regulations, and then further in an effort to understand how apps might be useful for architecture students studying active building or environmental systems. To simulate a typical educational or "street" situation, soundwalks are taken in which various sources are measured using apps on different phones as well as a Type 1 sound level meter for reference. The results are compared and a recommendation of appropriate application is made.

Session 2aNSb

Noise, Structural Acoustics and Vibration, and Architectural Acoustics: Noise and Vibration Impacts from Crossfit Training Facilities

Steve Pettyjohn, Cochair
The Acoustics & Vibration Group, Inc., 5765 9th Avenue, Sacramento, CA

James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Chair's Introduction—10:40

Invited Papers

10:45

2aNSb1. The effectiveness of resilient sports flooring on noise from crossfit activities. Matthew V. Golden and Paul Gartenburg (Pliteq, 1370 Don Mills Rd., Unit 300, Toronto, ON M3B 3N7, Canada, mgolden@pliteq.com)

Crossfit facilities in dense, multi-use facilities are growing in numbers every day. Even minimally sensitive receivers can be very significantly impacted by Crossfit activities. Consequently, vibration and vibration induced noise in structures as a result of heavy weight drops are a never ending issue. This paper compares data recorded from heavy weight drops in an ASTM E492 floor/ceiling testing suite. The resultant sound pressure level in the receiving room and vibration levels in the structure are measured. Several different standard and high performance resilient sports floorings are compared. These sports floors are tested on single concrete slabs as well as various floating slabs. The floating slabs include continuous rubber underlayments, discrete rubber isolators and spring jack-up floating floors. In addition, laboratory impact and airborne isolation performance results of a spring jack-up floating floor are reported.

11:05

2aNSb2. Mock-up testing for CrossFit vibration. David Manley and Ben Bridgewater (D.L. Adams Assoc., Inc., 1536 Ogden St., Denver, CO 80218, dmanley@dlaa.com)

A new construction mixed-use building was proposed in Crested Butte, CO, to consist of ground floor fitness, second floor commercial office, and third floor residential. It was anticipated that the ground floor fitness space would include CrossFit style workouts which often incorporate Olympic weight lifting and dropping activities. Due to the remote project location, five different mock-ups of impact and vibration isolation products were assembled in a gym in Denver to test the reduction in vibration levels from heavy weight impacts. Acceleration levels were measured on the concrete slab on grade close to the weight drop and in the soil outside the gym area. Mock-up testing methodology and results are presented. Based on the test results, recommendations on products, assemblies, and construction details were provided.

11:25

2aNSb3. Impact of heavy weight drops at Crossfit® training facilities on adjacent doctor's examination rooms and on conference rooms. Steve Pettyjohn (The Acoust. & Vib. Group, Inc., 5765 9th Ave., Sacramento, CA, spettyjohn@acousticsandvibration.com)

Significant sound and vibration are generated when members of Crossfit® training facilities drop heavy barbell weights to the floor from waist or shoulder height. Weights between 40 kg (90 lbs) and 100 kg (225 lbs) are repeatedly lifted from the floor to should height, then dropped to the floor and then picked up again. Crossfit® combines a variety of exercises using weights, pull up bars, and tires for dragging to increase the strength of participants. The sound and vibration generated by dropping of the weights is the only activity that has generated strong reaction from adjacent spaces. The Crossfit® facilities are found in many spaces, but often in warehouse/office or strip offices, i.e., locations with many types of tenants. Examples are Crossfit® facilities next to optometrists, conference rooms, and massage parlors. The sound and vibration generated by the weight drops and resulting sound and vibration in the receiver space are presented for several conditions of the equipment and the floor. Several options for reducing the transmission of excessive sound and vibration into adjacent spaces is presented. These include modifications to the weights, floor toppings, floor dimensions, and wall construction.

Contributed Paper

11:45

2aNSb4. Investigation of acoustical environment of multi-story buildings in Korea for applicability of EN 12354. Sung M. Kim, Hansol Lim, and Jin Y. Jeon (Architectural Eng., Hanyang Univ., Seongdong-gu Wangsimni-ro 222, Seoul 133-791, South Korea, rainbear0622@gmail.com)

In multi-story buildings, sound reduction can be evaluated using sound prediction model defined in the EN 12354 in Europe generally. The reduction model deals with the calculation and evaluation of direct and flanking transmission between two rooms through each structure elements for example the floor, ceiling, inner wall, or etc. In case of Korea, general residential type of is multi-story building so the noise transmission between neighbors is the most well-known acoustical problem. The EN 12354 can be considered as an important evaluation method, however the acoustic environment in Korea should be considered for the applicability of prediction model. Therefore, acoustical environments of multi-story buildings were investigated using field survey in order to evaluate the applicability through EN 12354.

TUESDAY MORNING, 24 MAY 2016 SALON H, 8:00 A.M. TO 11:45 A.M.

Session 2aPA

Physical Acoustics and Biomedical Acoustics: Vortex Beams and Radiation Torque Physics I

Philip L. Marston, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

2aPA1. Radiation stress, momentum, angular momentum, and power relationships for axisymmetric objects in acoustic vortex and coaxial wavefields. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Likun Zhang (Phys. Dept. & Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, Austin, TX)

Starting with P. Westervelt's formulation of the radiation stress tensor and G. Maidanik's application to angular momentum in the 1950s, some recent theorems concerning acoustic radiation forces, torques, angular momentum, and power flow will be illustrated. Applications include acoustic vortex wavefields, Bessel beams, and other invariant beams. While derivations are simplified for objects surrounded by inviscid fluids [L. Zhang and P. L. Marston, Phys. Rev. E. **84**, 035601 (2011); L. Zhang and P. L. Marston, Phys. Rev. E. **84**, 065601 (2011); L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **131**, EL329–EL335 (2012)], for spheres in slightly viscous liquids limiting approximations are known at long wavelengths that recover results derivable by other approaches [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **136**, 2917–2921 (2014); P. L. Marston, POMA **19**, 045005 (2013)]. Theorems obtained are helpful for relating negative radiation forces to the asymmetry of the scattering pattern for spheres in Bessel beams. The relation with extinction theorems is noted [L. Zhang and P. L. Marston, Bio. Opt. Express **4**, 1610–1617 (2013); (E) **4**, 2988 (2013)] as well as momentum and angular momentum radiated by sources. [Work supported in part by ONR.]

8:30

2aPA2. Wave vortices, structured beams, and their interaction with matter: What can we learn from analogies? Karen Volke-Sepulveda (Instituto de Fisica, Universidad Nacional Autonoma de Mexico, Apdo. Postal 20-364, Mexico City, Mexico D.F. 01000, Mexico, karen@fisica.unam.mx)

One of the many reasons making the study of waves so fascinating is the opportunity it brings to establish links among different areas of physics. In particular, the analogies between optics and acoustics have led to important advances in both directions. Fundamental research on structured wave fields and phase singularities, such as Bessel beams and helical waves, as well as their interaction with matter, is one of the areas that has been benefited greatly from these cooperative advances. Regarding topological studies of scalar wave fields, for instance, sound waves might represent a more versatile alternative than light, since their phase structure can be measured directly with respect to a reference signal. In contrast, the intensity distribution of light can be directly observed, whereas the intensity

distribution of an acoustic field has to be determined from measurements of the sound pressure. Therefore, optical and acoustical demonstrations complement and reinforce each other. In this presentation, some specific examples of this productive interchange will be discussed as a starting point, involving the study of acoustic vortices. This will pave the way for introducing some recent results on the analysis of structured light beams, which may find direct translation to the acoustic realm.

8:55

2aPA3. Angular momentum conservation and symmetry. Likun Zhang (Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu), Yong Li (CNRS-Université de Lorraine, Vandœuvre-lés-Nancy, France), Xue Jiang, Bin Liang, and Jian-chun Cheng (Nanjing Univ., Nanjing, China)

The conservation of angular momentum during the interactions of sound fields with objects is associated with the symmetry in the fields and the objects. Simple arguments of symmetry and/or conservation can have efficient applications in analyses of angular momentum transfer during the interactions. Scenarios that will be illustrated include: (a) conserved phase distribution for analysis of torques exerted by vortex beams on axis-symmetric objects for its connection with absorption [L. Zhang and P. L. Marston, JASA **129**, 1679–1680 (2011); Phys. Rev. E. **84**, 065601 (2011)], (b) symmetric superposition of vortex beams for analysis of torques exerted by orthogonal waves on a small compressible object in a slightly viscous fluid [L. Zhang and P. L. Marston, JASA **136**, 2917–2921 (2014)], and (c) broken symmetry in a planar meta-screen for introducing angular momentum into a plane sound field of null angular momentum [Y. Li *et al*., Phys. Rev. Appl. **4**, 024003 (2015); X. Jiang *et al*., in preparation].

Contributed Paper

9:20

2aPA4. Vortex beams and radiation torque for kidney stone management. Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Michael Bailey, Bryan W. Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, mike.bailey.apl@gmail.com), M. Terzi (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Seattle, Washington), A. Nikolaeva, S. Tsysar, and Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Our team previously developed an instrument to reposition kidney stones with acoustic radiation force. In a clinical trial, the technology was used to transcutaneously facilitate passage of small stones and to relieve pain by dislodging obstructing large stones. Acoustic trapping and manipulation of kidney stones in water has recently been investigated using both single element and sector arrays in the range of 0.3–1.5 MHz. Experimental holographic reconstruction of the transducer surface velocity confirmed the proper operation of each transducer. Human stones approximately 5 mm, as well as glass and aluminum beads, were placed on a flat tissue phantom in a water bath. During exposure, stones were drawn to the beam axis, and then controllably translated along the surface in any direction transverse to the beam. The phase between sector elements could be used to control the vortex size, as well as rate and direction of rotation of the trapped object. The trapping effect was disrupted at increased transducer output, possibly by generation of acoustic streaming. In conclusion, a method was tested for transverse acoustic trapping of kidney stones with vortex beams. [This work was supported by RBBR 14-02-00426, NIH NIDDK DK43881, DK104854, and DK092197, and NSBRI through NASA NCC 9-58.]

Invited Papers

9:35

2aPA5. Characterization of phased array-steered acoustic vortex beams. Jhon F. Pazos-Ospina (School of Mech. Eng., Universidad del Valle, Cali, Valle del Cauca, Colombia), Ediguer E. Franco (Universidad Autónoma de Occidente, Cali, Colombia), and Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Acoust. and Vib. Lab., Ciudad Universitaria Meléndez. Bldg. 351., Cali, Valle del Cauca 760032, Colombia, joao.ealo@correounivalle.edu.co)

Acoustic vortex (AV) beams generation is a subject of current interest. Even though different applications have been proposed using AV, their potential of use is still to be explored. Recent research works on particle manipulation use phased array systems for AV generation because it allows a flexible beam configuration, i.e., the beam can be easily focalized and modified in its shape. However, little attention has been paid to the fact that the AV can also be electronically steered. In view of this, this work presents an study of the steering capability of an AV. In particular, we analyze the effect of the applied delay law on the structure of AV beams steered at different angles using an array transducer of 32 equidistant elements, deployed on a triangular lattice, operating at 40 kHz. Special attention is paid to the appearance of grating vortices and their characteristics. The effect of the individual element directivity on the resultant beam is also studied. Experimental measurements were carried out in order to validate numerical estimations. Obtained results paves the way for the use of electronically steered vortices in different applications. Also, the potential of use of acoustic grating vortices is discussed.

10:00–10:15 Break

10:15

2aPA6. Sonic Screwdrivers and Tractor Beams: what the Acoustics Community often gets *wrong* about gradient forces versus radiation pressure. Gabriel C. Spalding (Phys., Illinois Wesleyan Univ., 201 E. Beecher St., Bloomington, IL 61701-7222, gspaldin@iwu.edu), Patrick Dahl (Phys., Illinois Wesleyan Univ., Evanston, IL), Zhengyi Yang (Inst. for Medical Sci. & Technol., Univ. of Dundee, St. Andrews, United Kingdom), Peter Glynne-Jones (Eng. Sci., Univ. of Southampton, Southampton, United Kingdom), Michael P. MacDonald (Inst. for Medical Sci. & Technol., Univ. of Dundee, Dundee, United Kingdom), Christine Demore, and Sandy Cochran (School of Eng., Univ. of Glasgow, Glasgow, United Kingdom)

Through discussion of our sonic screwdriver and tractor beam experiments, we aim to highlight the respective advantages of conservative and non-conservative forces. Commonly in acoustic trapping, conservative, gradient-induced mechanisms (e.g., standing waves) are used to manipulate matter. Such situations are reasonably described in terms of potential energy landscapes, an approach also applied to optics, for applications such as cell sorting [MacDonald et al, Nature 426 (2003)]. No such description is possible for radiation pressure, which is non-conservative, a distinction that is sometimes muddled in the literature, although it was made clear even in early work [e.g., King, Proc. R. Soc. Lond. A **147** (1934); Gor'kov, Sov. Phys. Doklady **6**, 773 (1962)]. Our "sonic screwdriver" makes use of two non-conservative mechanisms: levitation by radiation pressure and rotation by transfer of azimuthal momentum components [Demore *et al*., PRL **108** (2012)]. We also note that the term "tractor beam" has often been reserved to describe an effect involving non-conservative forces, and demonstrate an attractive force produced in such an arrangement, even against a net momentum flux [Demore *et al*., PRL **112** (2014)].

10:40

2aPA7. Single-beam acoustical tweezers: Concept, theory, and experimental realization. Diego Baresch (Institut des NanoSci. de Paris, UPMC, 4, Pl. Jussieu, Paris 75005, France, diego.baresch@upmc.fr), Régis Marchiano (Institut Jean Le Rond D'Alembert, UPMC, Paris, France), and Jean-Louis Thomas (Institut des NanoSci. de Paris, UPMC, Paris, France)

Vortex beams are characterized by wavefronts twisting around their axis of propagation. A three dimensional region of silence tightly bounded by a high intensity ring emerges from the phase's screw dislocation, which can be used for particle manipulation. In this paper, we present the first experimental realization of selective 3D acoustical tweezers with a single beam. It had long been recognized that acoustic radiation pressure pushed objects in the direction of propagation. As we will show, by accurately modeling the radiation pressure exerted on an elastic sphere lying in any arbitrary location, it is possible to identify this axial ejection and to propose a solution to counterbalance this effect. The acoustic puling force of a properly focused vortex beam is explained as a result of the particle's monopolar mode annihilation on the propagation axis. Consequently, these beams act as a stable trap in three dimensions. An experimental realization has been proposed in water using a ultrasonic beam (1 MHz). Elastic particles in the sub-millimetric range have been accurately trapped and manipulated. This setup's dexterity and selectivity is the macroscopic analogous of optical tweezers exerting, however, forces which are 5 orders of magnitude stronger at comparable intensities.

11:05

2aPA8. Taming tornadoes: Controlling orbits inside acoustic vortex traps. Asier Marzo, Mihai Caleap, and Bruce W. Drinkwater (Mech. Eng., Univ. of Bristol, University Walk, Queen's Bldg. 1.1, Bristol, Bristol BS8 1TR, United Kingdom, amarzo@hotmail.com)

Bessel-shaped acoustic vortices have been theorized to be ideal candidates for tractor beams—that is, acoustic beams that exert forces opposite to the direction of propagation. Recently, it has been shown experimentally that these pulling forces exist and that particles can be trapped inside a focused vortex beam, both in water and air. However, our experiments and simulations reveal that the behavior of particles trapped inside a focused vortex is strongly dependent on the particle size. We trapped expanded polystyrene spheres (ranging from 0.6 mm to 4 mm diameter) 4 cm above a phased array made of 52 transducers (40 kHz and 1 cm diameter each) that generated an airborne acoustic vortex. Depending on its size, a particle remained trapped, oscillated between being trapped and orbiting, orbited or was ejected; only small particles (< 1 mm) were stably trapped. We then devised a new procedure for making vortex traps stable for a wider range of particle sizes; the strategy relies on switching the direction of the vortex at 4 kHz. In addition, by regulating the amount of time that each direction is emitted, it is possible to controllably rotate symmetric particles. This enables new applications in particle manipulation such as centrifugation of cells or actuation of micromachines.

Contributed Paper

11:30

2aPA9. Helicity sensitive processing and imaging of a first order vortex beam scattered by a sphere. Viktor Bollen (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu), Daniel S. Plotnick (Acoust., Appl. Res. Lab, Seattle, WA), and Philip L. Marston (Washington State Univ., Pullman, WA)

A vortex beam's wavefield has a null on the axis of propagation and an angular phase ramp proportional to the order of the beam. The rapid phase ramp at the null leads to the possibility of high-resolution imaging and precise alignment of the system. Using a modified four-panel transducer, a first order vortex beam was generated by driving each panel with an appropriate phase shift [V. Bollen *et al*., Proc. Meet. Acoust. **19**, 070075 (2013)]. Utilizing this transducer, a solid sphere was insonified and the backscattering measured. Recording the backscattering on each panel separately allowed selection between helicity neutral and helicity sensitive detection modes, without changing the experimental setup, by introducing individual phase shifts in post-processing [V. Bollen, *et al*., J. Acoust. Soc. Am. **137**, 2439 (2015)]. Using time delay-and-sum imaging algorithms, we created high-resolution three-dimensional profiles of the beam, relating the sphere location to the beam pattern. With cross-correlation involving measured and computed scattering amplitudes, we showed agreement with a Kirchhoff integration based simulation of the beam. [Work supported by ONR.]

Session 2aPP

Psychological and Physiological Acoustics and Signal Processing in Acoustics: Approaches to Improve Speech Understanding in Noise

Eric Healy, Cochair
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Signal Processing Reseearch, Starkey Hearing Technologies, 8602 Zachman Circle, Eden Prairie, MN 55344

Chair's Introduction—8:00

Invited Papers

8:05

2aPP1. Understanding the speech-understanding problems of older adults. Larry E. Humes (Dept Speech & Hearing Sci., Indiana Univ., Bloomington, IN 47405-7002, humes@indiana.edu)

This presentation will provide an overview of the nature of the difficulties experienced by older adults with impaired hearing when listening to speech in noise. Many older adults may experience a "triple whammy" of difficulties: (1) peripheral pathology in the cochlea or auditory nerve; (2) age-related deficits in central-auditory function; and (3) age-related changes in cognition that impact speech perception. Each of these difficulties may appear in isolation or in various combinations for a given older adult. This presentation will review the evidence surrounding each of these potential sources of difficulty and the relative contribution of each across various types of "noise" backgrounds. In addition, the relative importance of each source of difficulty for aided and unaided listening conditions will be considered. [Work supported, in part, by NIA R01 AG008293.]

8:25

2aPP2. Interactions between spectral and temporal processing on speech understanding in cochlear-implant users. Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Understanding speech in noise remains one of the greatest challenges facing people with cochlear implants (CIs). For normal-hearing listeners, speech understanding in noise seems limited to a large extent by the modulation energy (temporal fluctuations) present in the noise, rather than the noise energy itself. This finding does not seem to apply in CI users, for whom no difference is observed between maskers with or without inherent temporal fluctuations. The current explanation for this qualitative difference between the results from normal-hearing listeners and CI users is based on the lack of spectral resolution in CIs, which results in an overlap of stimulation from adjacent channels, and a resultant smoothing of the temporal envelopes produced by noise. This study reviews evidence for this hypothesis, and tests some arising predictions, including the effects of narrower stimulation patterns (tripolar versus monopolar) and increased spacing between active electrodes in both CI users and normal-hearing listeners using envelope-vocoder simulations of CI processing. [Work supported by NIH grant R01 DC012262.]

8:45

2aPP3. Efficacy and viability of an algorithm to improve speech understanding in noise for the hearing impaired. Eric W. Healy (Speech & Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu)

A primary complaint of hearing-impaired individuals involves poor speech understanding when background noise is present. Hearing aids and cochlear implants often allow good speech understanding in quiet backgrounds. However, the listeners' noise intolerance and the devices' inability to effectively combat background noise often conspire to produce poor performance in noise. Perhaps surprisingly, effective solutions to this problem have remained elusive despite considerable effort. One promising solution involves a single-microphone algorithm to extract speech from background noise. The algorithm is based on the concept of the ideal binary or ratio mask, and employs standard machine-learning techniques to train a deep neural network to estimate the mask, given only the speech-plus-noise mixture. Existing data indicate that large intelligibility increases by hearing-impaired listeners may be obtained across a variety of noisy conditions. In this talk, an overview of this approach will be provided, and the potential for implementation into hearing aids and cochlear implants will be discussed. [Work supported by NIH.]

9:05

2aPP4. Enhancement of spectral contrast and spectral changes as approaches to improving the intelligibility of speech in background sounds. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Cochlear hearing loss is accompanied by reduced frequency selectivity, which contributes to problems in understanding speech in background sounds. The excitation pattern evoked by sounds like vowels is "flatter" in an impaired ear than in a normal ear. Two approaches to partially compensating for the effects of reduced frequency selectivity are described. The first uses spectral contrast enhancement on a frame-by-frame basis. Only features of the short-term spectrum that would be represented in a normal ear are enhanced. Experimental evaluations of this approach showed that it could improve the intelligibility of speech in noise by a small amount, but that this required a fine frequency analysis, in turn requiring relatively long frames and time delays that might be too long for use of the method in hearing aids. A second approach enhances spectral changes over time, based on the idea that information is speech is carried by spectral changes. This approach also led to modest improvements in the intelligibility of speech in noise, especially when the amount of enhancement was made to vary with frequency depending on the hearing loss at that frequency and when the parameters of the processing were selected for each individual using a genetic algorithm.

9:25

2aPP5. Metrics for evaluating noise suppression in hearing aids. James Kates (Speech Lang. Hearing Sci., Univ. of Colorado., 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

Noise suppression in audio systems such as hearing aids typically involves modifying the speech envelope. Noisy speech segments having poor signal-to-noise ratios are attenuated while those considered to be primarily speech are left at or near full intensity. Both spectral subtraction and ideal binary mask noise suppression work on this principle. Recent work in the development of speech intelligibility and speech quality metrics indicates that the envelope fidelity, computed as the correlation coefficient between the processed noisy speech envelope and the original noise-free speech envelope, is an important factor in determining noise-suppression benefit. This result implies that the best noise suppression system is one in which the envelope of the processed noisy speech is restored to match as closely as possible that of the original noise-free speech. The potential benefits and limitations of envelope restoration for noisy speech are analyzed using the HASPI intelligibility and HASQI quality metrics, and the performance of existing noise suppression processing is compared to that of envelope restoration for both normal-hearing and hearing-impaired listeners.

9:45–10:00 Break

10:00

2aPP6. A novel binaural speech enhancement algorithm for hearing impaired listeners. Tao Zhang (Signal Processing Res., Starkey Hearing Technologies, 8602 Zachman Circle, Eden Prairie, MN 55344, tzhang28@ieee.org)

With the advent of wireless ear to ear communication, binaural speech enhancement becomes feasible in commercial products such as hearing aids. The first such an example is a fixed binaural beam former that attenuates noise from the nonlook direction while preserving the speech in the look direction. While such an algorithm has shown significant benefits in the lab, it has not produced the wow effect in the field as we have all expected. In this paper, we will discuss some of the practical issues limiting the benefits of such an algorithm in the field. In addition, a novel binaural speech enhancement algorithm is proposed by balancing robustness along with optimal performance. Its benefits will be demonstrated by comparing it with the leading binaural speech enhancement algorithms. Furthermore, future research directions will be discussed.

10:20

2aPP7. Comprehensive evaluation of binaural hearing aid pre-processing strategies—Speech intelligibility in realistic noise scenarios. Stephan Ernst, Regina Baumgärtel, Christoph Völker, Anna Warzybok, Mathias Dietz, Volker Hohmann, and Birger Kollmeier (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Carl von Ossietzky Universität Oldenburg, Oldenburg 26111, Germany, stephan.ernst2@uni-oldenburg.de)

For the development of new signal processing approaches, e.g., for hearing aids, a final stage of evaluation is crucial. However, the evaluation procedures for assessing performance and benefit often represent a vast and not unified realm. Thus, we suggest a comprehensive evaluation to describe the efficacy, i.e., the anticipated real-world benefit as precise as possible. This comprises physical, instrumental, and perceptual (human) measurements of objective measures, as well as subjective attributes. Eight signal pre-processing strategies were evaluated following this concept, including directional microphones, coherence filters, single-channel noise reduction, binaural beamformers, and their combinations. Speech reception thresholds (SRTs) were measured with normal-hearing and hearing-impaired listeners in three realistic noise scenarios and compared with predictions of common instrumental measures. Although hearing-impaired listeners required a better signal-to-noise ratio to obtain 50% intelligibility than listeners with normal hearing, no differences in SRT benefit (of up to 4.8 dB) from the different algorithms were found between the two groups. This suggests a possible application of noise reduction schemes for listeners with different hearing status. Although the instrumental measures can predict the individual SRTs without pre-processing, development is necessary to predict the benefits obtained from the algorithms at an individual level.

10:40

2aPP8. How individual differences in sensory coding and attentional control impact understanding speech in noise. Barbara Shinn-Cunningham (Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu), Dorea Ruggles (Psych., Univ. of Minnesota, Minneapolis, MN), Inyong Choi (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), Hari Bharadwaj (Athinoula A. Martinos Ctr. for Biomedical Imaging, Massachusetts General Hospital, Boston, MA), Golbarg Mehraei (Elec. Eng., Tech. Univ. of Denmark, Lyngby, Denmark), and Lengshi Dai (Biomedical Eng., Boston Univ., Boston, MA)

Historically, the majority of psychoacoustic studies of hearing ability have viewed individual differences as noise: a nuisance that makes it difficult to see the effects that different acoustic conditions have on auditory perception. This talk reviews how we have begun to use individual differences to tease apart the processes that affect perception, with a particular focus on how listeners understand speech when there are competing sound sources. We find that individual subjects show consistent differences in their ability to understand speech in noise. These consistent differences can come both from differences in the fidelity of sensory coding and from differences in the ability to focus selectively on important sound and suppress unimportant sound. Importantly, which of these factors predicts performance depends greatly on the details of the stimuli used in a given task, and what stage of processing is the resulting bottleneck, determining performance. When fine differences in the sound content, such as small differences in location or pitch, are critical for a task, differences in sensory fidelity dominate individual differences in ability. However, when the acoustic features important for separating sound streams and identifying the target stream from the mixture are very distinct, individual differences in ability reflect differences in attentional control. These results highlight how understanding speech in noise depends on complex interactions between the ear and the brain.

11:00

2aPP9. The "meaning" in noise—Evidence for bottom-up information masking of within channel modulation coding of speech. Simon Carlile (Starkey Hearing Res. Ctr., F13, Anderson Stuart Bldg., Camperdown, Sydney, New South Wales 2006, Australia, simonc@physiol.usyd.edu.au)

Spectro-temporal variations are a consequence of dynamic structural variations of a sounding body. The human auditory system is highly optimized for the detection, segregation and analysis of one class of such variations—speech. Here, we will examine some consequences of this optimisation in the context of complex listening involving multiple concurrent talkers. In such conversational settings, listeners rapidly shift their attention from one to another talker so foreground and background are defined dynamically by listener intent. Up-regulation of the attended-to talker is generally thought to result from endogenous, top-down attention. Here, we review a recent report indicating that substantial informational masking between concurrent talkers may result from bottom-up interactions between sources within frequency modulation channels. This indicates that temporally dynamic aspects of within-frequency channel processing plays a very important role in speech masking by the sort of "noise" most commonly encountered in natural conversational settings. Possibly even more surprising, these interactions appear to be modulated by spatial attention. Attentional enhancement of a foreground sound might then also involve processes at the level of the within-frequency channel. Whether this occurs prior to or as a consequence of grouping is a question of some functional significance but also points to the importance of knowing the listeners focus of attention.

11:20

2aPP10. Neural representations of speech, and speech in noise, in human auditory cortex. Jonathan Z. Simon (Dept. of Elec. & Comput. Eng., College Park, MD 20742, jzsimon@umd.edu)

We investigate how continuous speech, whether presented alone, degraded with noise, or masked by other speech signals, is represented in human auditory cortex. We use magnetoencephalography (MEG) to record the neural responses of listeners to continuous speech in a variety of contexts. We find that cortical representations of continuous speech are robust to noise under a wide variety of conditions, including clean speech, additive stationary noise, band-vocoded speech and noise, and speech with competing speakers. In the last case, individual neural representations of the speech of both the foreground and background speaker are observed, with each being selectively phase locked to the rhythm of the corresponding speech stream. In all observed cases, the temporal envelope of the acoustic speech stream can be reconstructed from the observed neural response to the speech.

11:40

2aPP11. Neuro-steered noise suppression for auditory prostheses. Tom Francart, Neetha Das (NeuroSci., KU Leuven, Herestraat 49 Bus 721, Leuven 3000, Belgium, tom.francart@med.kuleuven.be), Simon Van Eyndhoven, Wouter Biesmans, and Alexander Bertrand (Elec. Eng., KU Leuven, Leuven, Belgium)

In a multi-speaker scenario, a major challenge for noise suppression systems in hearing instruments is to determine which sound source the listener is attending to. It has been shown that a linear decoder can extract a neural signal from EEG recordings that is better correlated with the envelope of the attended speech signal than with the envelopes of the other signals. This can be exploited to perform auditory attention detection (AAD), which can then steer a noise suppression algorithm. The speech signal is passed through a model of the auditory periphery before extracting its envelope. We compared 7 different periphery models and found that best AAD performance was obtained with a gamma-tone filter bank followed by power-law compression. Most AAD studies so far have employed a dichotic paradigm, wherein each ear receives a separate speech stream. We compared this to a more realistic setup where speech was simulated to originate form two different spatial locations, and found that although listening conditions were harder, AAD performance was better than for the dichotic setup. Finally, we designed a neuro-steered denoising algorithm that informs the voice activity detection stage of a multi-channel Wiener filter based on AAD, and found a large signal-to-noise-ratio improvement at the output.

Session 2aSC

Speech Communication: Intelligibility, Hearing Impairment, Aging (Poster Session)

Ewa Jacewicz, Chair
Department and Speech and Hearing Science, The Ohio State University, 1070 Carmack Road, 110 Pressey Hall, Columbus, OH 43210

All posters will be on display from 8:00 a.m. to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:30 a.m., and authors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m.

Contributed Papers

2aSC1. Band importance functions of listeners with cochlear implants. Adam K. Bosen and Monita Chatterjee (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, adam.bosen@boystown.org)

Band Importance Functions (BIFs) are measured by dividing the acoustic spectrum into discrete bands and estimating the relative contribution of each band to speech intelligibility. Previous studies have demonstrated strong across-subject consistency for normal hearing (NH) listener BIFs. However, it is unlikely that cochlear implant (CI) listeners will demonstrate similar consistency, because many physiological and psychophysical properties of hearing vary across CI listeners and across electrodes within an ear. Here, we measured BIFs by testing speech intelligibility of IEEE sentences filtered to contain a random subset of bands from a listener's clinical MAP and regressing across trials to determine each band's average contribution. BIFs obtained from NH listeners with vocoded speech that either did or did not simulate cross-band interaction followed the characteristic inverted "U" shape with a peak around 1–2 kHz that has been previously observed. In comparison, CI listeners had BIFs that were less similar across listeners and across ears within the same listener. These results indicate that CI listeners use idiosyncratic listening strategies for speech perception that are ear-specific and cannot be fully accounted for by cochlear spread of excitation.

2aSC2. Intelligibility of British English for American younger and older adults with and without hearing loss. Caroline Champougny, Sarah H. Ferguson, and Sadie Schilaty (Commun. Sci. and Disord., Univ. of Utah, 390 S 1530 E, Rm. 1201 BEH SCI, Salt Lake City, UT 84112, u6002580@utah.edu)

In clinics, patients with hearing loss often report that they have difficulty understanding British-accented speech, despite having in the past watched and understood British television shows without trouble. Clopper and Bradlow (2008) examined the intelligibility of four United States regional dialects by American listeners, but to our knowledge, no study to date has quantified the intelligibility of non-U.S. varieties of English for American listeners. The present study will test the intelligibility of British English sentences by American younger and older adult listeners with normal hearing as well as a group of older adults with hearing loss. Participants will be presented with Basic English Lexicon (BEL) sentences produced by one American and one British talker; sentences will be presented in quiet and in a background of 12-talker babble. The resulting data will reveal how well hearing-impaired older adults understand British English, whether talker accent interacts with the presence or absence of noise, and whether older adults are disproportionately impaired in understanding British English compared to younger adults.

2aSC3. Perceptual speech intelligibility and speech production variability in Mandarin-speaking children with cerebral palsy. Li-mei Chen, Yu Ching Lin (Dept. of Foreign Lang., National Cheng Kung Univ., 1 University Rd., Tainan 701, Taiwan, leemay@mail.ncku.edu.tw), Katherine C. Hustad, and Raymond D. Kent (Waisman Ctr., Madison, WI)

Perceptual speech intelligibility index can provide an objective standard for evaluating how speech deficiencies affect listeners' judgments. This study investigated the effective variables for assessing speech intelligibility in Mandarin-speaking children with cerebral palsy. Acoustic measurements include vowel working space, vowel duration, pitch variation, and intensity variation of speech samples collected from picture naming tasks in two children with cerebral palsy (CP) and two typically developing children (TD) at four years old. Only clear word productions in terms of perceptual judgment and vowel formant display were incorporated to secure the reliability of data analysis. For speech intelligibility, eight judges were recruited to record the words they heard from the speech samples. Major findings are: (1) CP children had a smaller vowel space area due to limited control of tongue compared to TD children; (2) CP children showed longer vowel duration because they spent more time producing speech due to limited motor control; (3) CP children exhibited greater pitch and intensity variations due to instability in speech production. More data from more participants should be included for analysis to verify the preliminary findings.

2aSC4. Age-related differences in talker segregation and selection: Contributions of attention to voice features. William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, Medical University of South Carolina, 135 Rutledge Ave., MSC 550, Charleston, SC 29425, bologna@musc.edu), Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Focused attention on expected voice features, such as F0 and spectral envelope, may facilitate segregation and selection of a target talker in competing talker backgrounds. Age-related declines in attention may limit these abilities in older adults, resulting in poorer speech understanding in complex environments. To test this hypothesis, younger and older adults with normal hearing listened to sentences with a single competing talker. For a majority of trials, listener attention was directed to the target by a cue phrase that matched the target talker's F0 and spectral envelope; on these trials, younger adults outperformed older adults. On the remaining "probe" trials, the target's voice unexpectedly differed from the cue phrase in terms of F0 and spectral envelope; performance declined for both groups, and younger and older adults performed similarly. Thus, older adults performed poorer than younger adults only when attention could be focused on an expected voice. Moreover, older adults responded more frequently than younger adults with words from the competing sentence rather than the target sentence. Taken together, these results support the hypothesis that declines in the ability to focus attention on expected voice features contribute to speech understanding difficulties of older adults in complex environments. [Work supported by NIH/NIDCD.]

2aSC5. Speech produced in noise: Relationship between listening difficulty and spectral and durational parameters. Simone Graetzer, Pasquale Bottalico, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, sgraetz@msu.edu)

Conversational speech produced in noise can be characterized by increases in intelligibility relative to conversational speech produced in quiet. The objectives of the study were to evaluate the listening difficulty of speech produced in different noise and style conditions, to evaluate the spectral and durational speech modifications that occurred in these conditions, and to determine whether the spectral or durational parameters predicted listening difficulty. Nineteen subjects were instructed to speak at normal or loud volumes in the presence of background noise at 40.5 dB(A) and babble noise at 61 dB(A). The speech signal was amplitude-normalised, combined with pink noise to obtain a signal-to-noise ratio of −6 dB, and presented to 20 raters who judged their listening difficulty. It was found that vowel duration, fundamental frequency (f_o, in semitones) and the proportion of the spectral energy in high relative to low frequencies increased with the level of the noise, independently of the effect of style. Listening difficulty was lowest when the speech was produced in the presence of high level noise and in the loud style, indicating improved intelligibility. The difference in spectral energy was observed to predict listening difficulty, and, therefore, intelligibility scores (*IS*, the percentage of words understood correctly).

2aSC6. Effect of auditory-motor mapping training and speech repetition training on consonant and vowel accuracy in minimally verbal children with autism spectrum disorder. Karen V. Chenausky, Andrea Norton, and Gottfried Schlaug (Neurology, Beth Israel-Deaconess Medical Ctr., Beth Israel-Deaconess Medical Ctr., 330 Brookline Ave., Boston, MA 02215, kvchenau@bidmc.harvard.edu)

Various therapies exist for teaching first words to minimally verbal (MV) children with autism spectrum disorder (ASD). Previous outcome measures have focused on number of words imitated or produced spontaneously, or on communication rate. No studies thus far have examined phonetic accuracy in MV ASD as a result of therapy, yet understanding whether therapy improves articulation in MV ASD is important for understanding how speech is affected in ASD and how best to treat these children. In this preliminary study of 30 children with MV ASD, we report on perceptual analysis of their speech after 25 sessions of one of two therapies, employing bisyllabic stimuli with a variety of consonant types. Twenty-three children received Auditory Motor Mapping Training (AMMT), where stimuli are intoned at approximately one syllable per second. Seven children, matched on age and cognition to 7 AMMT participants, received Speech Repetition Training (SRT), where non-intoned stimuli are presented at a normal speech rate. ANOVAs indicated that AMMT resulted in greater gains in consonant production accuracy than SRT. Both therapies produced equally significant improvement in vowel accuracy. Additional, ongoing analyses concern participants' accuracy for different consonant types, investigating whether accuracy is greater for earlier-appearing than later-appearing consonant types.

2aSC7. A longitudinal study of the effects of aging on speech breathing: Evidence of decreased expiratory volume in speech recordings. Simone Graetzer and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, sgraetz@msu.edu)

Age-related changes occur in speech that are associated with structural, physiological, and immunological processes involving the oral and nasal cavities, the larynx and pharynx, and the respiratory system. With aging, laryngeal tissues tend to degenerate or atrophy and laryngeal cartilages tend to ossify. These changes can lead to increased instability and perceived hoarseness or harshness, reduced loudness, and changes in fundamental frequency. In the respiratory system, a decline in lung and diaphragm elasticity and muscle strength can occur, and the thoracic cage can stiffen, leading to reductions in lung pressure and forced expiratory volume (hence, an increase in residual lung volume). In this study, recordings of three female and three male subjects were analyzed. These were made over the course of between 18 and 48 years (with a mean of 32.5 years). Samples of five minutes in length were extracted from each recording. Subsequently, trained raters measured the durations of exhalations during speech (termed "breath groups"). The results indicate decreases in breath group duration for most subjects as their age increased (especially from 65 years onwards), consistent with the decline in expiratory volume reported in the literature.

2aSC8. Evaluating how fine-grained changes in the spatial and temporal properties of audiovisual speech influence the perception of linguistic meter. Robert A. Fuhrman, Stanislaw Nowak, and Eric Vatikiotis-Bateson (Linguist, Univ. Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, robert.a.fuhrman@gmail.com)

The visual information that contributes to speech perception has been known for over a half century (Sumby and Pollack, JASA, 1954). More recent work has demonstrated two important constraints on the benefit of visual information to speech perception. First, the benefit patterns with temporal constraints on general audiovisual sensory processing, and degrades when auditory and visual signals are misaligned by an offset on the order of the duration of a syllable. Second, studies have demonstrated that low spatiotemporal frequency visual information is sufficient for boosting speech intelligibility. Collectively, these findings suggest that visual information facilitates processing of linguistic information organized in the speech stream primarily at the level of the syllable. Our experiments address how manipulation of the phasing and amplitude of visual components of the audiovisual stream associated with reiterant speech affects the perception of linguistic stress. Our analyses focus on determining the sensitivity of speech perception to changes in the fine-grained structure of visible motion (kinematics) and its alignment with the speech acoustics.

2aSC9. Vocal matching in interactions between mothers and their normal-hearing and hearing-impaired twins. Maria V. Kondaurova (Psychol. and Brain Sci., Univ. of Louisville, 699 Riley Hospital Dr. – RR044, Indianapolis, IN 46202, maria.kondaurova@louisville.edu), Laura C. Dilley (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Tonya R. Bergeson-Dana (Otolaryngol. – Head & Neck Surgery, Indiana Univ. School of Medicine, Indianapolis, IN), and Mary K. Fagan (Commun. Sci. and Disord., Univ. of Missouri, Columbia, MO)

Vocal matching, the ability to imitate phonetic properties of speech, was examined in interactions between mothers and their normal-hearing (NH) and hearing-impaired twins who used hearing aids (HAs) or a cochlear implant (CI). Vocalizations of three mother-twin triads were recorded in three sessions over 12 months. In one triad, the twins were 15.8 months old and NH. In another triad, the twins were 11.8 months; one was NH while the other had HAs. In the third triad, both twins were 14.8 months; one was NH while the other had a CI. A vocal match was defined as an instance of perceptual and acoustic similarity between adjacent maternal and infant utterances in relation to pitch height and contour, utterance duration, rhythm, or vowels and consonants. Reciprocal vocal matching occurred in 28% to 38% of infant vocalizations across triads. At session three, CI and HA infants' reciprocal vocal matches increased compared to two previous sessions and to those of NH siblings; reciprocal vocal matches in the NH dyad decreased over time. The results suggest that vocal matching is a part of linguistic interactions between mothers and their NH and HI infants and that pediatric hearing loss affects mothers' and infants' imitative abilities.

2aSC10. Clear speech benefits for perception of center-only and edge-only syllables. Jenna S. Luque, Nathan Maxfield, Jennifer J. Lister, and Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, jennaluque@usf.edu)

Clear speech is a speaking style that has been shown to enhance intelligibility in noise; however, the underlying reasons for this enhancement are less well understood. One hypothesis is that listeners require less acoustic information to identify syllables spoken in clear than in conversational speech, allowing them to make use of briefer dips in fluctuating noise. The present study tests this hypothesis by presenting six /bVd/ syllables ("bead, bid, bed, bayed, bad," and "bod"), produced in clear and conversational speech styles, to 20 monolingual native English-speaking listeners in a six-alternative forced-choice task. These syllables were modified to present

varying portions of the syllable to listeners (20, 40, 60, or 80 ms of the syllable preserved). First, center-only syllables were created, in which acoustic information around the vowel midpoint was preserved. Second, edge-only syllables were created, in which information from the vowel center was silenced and formant transitions preserved, with vowel duration either maintained or equalized. Preliminary results show some clear-speech benefits in the partial-syllable conditions, with benefits mainly at shorter gates for center-only syllables and mainly at longer gates for edge-only syllables. Implications for understanding the source of clear-speech benefits in noise and phoneme perception more generally will be discussed.

2aSC11. The effect of talker and listener depressive symptoms on speech intelligibility. Hoyoung Yi (Commun. Sci. and Disord., The Univ. of Texas at Austin, 2504A Whitis Ave., Austin, TX 78712-0114, hoyoung@utexas.edu), Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, Austin, TX), and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

It is widely recognized that depression is associated with deficits in communication behaviors (Miller, 1975), but few studies have examined speech intelligibility in talkers and listeners with elevated depressive symptoms. The present study examined intelligibility of conversational and clear speech sentences in the presence of speech shaped noise and one-talker babble noise. Talkers and listeners were young adults varying on the extent of depressive symptoms, and classified as having high depressive (HD) symptoms or low depressive (LD) symptoms based on a self-report scale (Radloff, 1977). The results showed that increased intelligibility through conversational-to-clear speech modifications was smaller for HD listeners than LD listeners during only one-talker babble noise which is consistent with Chandrasekaran *et al*. (2014). Intelligibility was scored lower for the sentences produced by HD speakers compared to LD speakers. Acoustic analyses indicate smaller conversation-to-clear speech modifications for speaking rate, energy in the 1-to-3 kHz, F0 mean, and F0 range for HD talkers compared to LD talkers. The results revealed that individuals with elevated depressive symptoms exhibit difficulty producing listener-oriented speaking style changes resulting in lower overall intelligibility. Adverse listening conditions involving informational masking, oft-present in typical social environs, present particularly challenging communication conditions for HD listeners.

2aSC12. The role of the carrier waveform in vocoded Mandarin speech perception with applications to cochlear implants. Jessica H. Schiltz (Speech & Hearing Sci., Arizona State Univ., 297 E LaSalle Ave., Apt. #: 205A, South Bend, IN 46617, jschiltz@nd.edu) and Visar Berisha (Speech & Hearing Sci., Arizona State Univ., Tempe, Arizona)

Cochlear implants (CI) enable patients to perceive sound by the generation and transmission of electrical impulses. Voice encoded (vocoded) audio is commonly used as a simulator of CI speech. Past studies of vocoded English concluded that coded speech intelligibility is independent of whether a sinusoidal or noise-based carrier is used. Based on this work, recent CI research for tonal languages like Mandarin, have opted for a sinusoidal carrier without considering the impact of the carrier on intelligibility. The importance of spectral cues in Mandarin speech necessitates further analysis of the relationship between the carrier and intelligibility. This study explored intelligibility differences between English and Mandarin vocoded speech. This approach assessed speech recognition of randomly presented phrases to normal hearing English and Mandarin listeners. Available frequency channels and carrier type were varied to compare their effects on Mandarin word and tone identification. Results indicated that audio processed with a sinusoidal carrier led to significantly lower Mandarin intelligibility scores. In comparison, no effect was observed between English and Mandarin intelligibility when a noise-based carrier was used. The data suggest that the nature of the carrier type affects tonal language intelligibility and warrants further research as an experimental consideration in vocoded speech studies.

2aSC13. Exploring the effects of masker spectro-temporal coherence on the informational masking of speech. Brian Roberts and Robert J. Summers (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk)

The effect of an extraneous formant on intelligibility is influenced by the depth of variation in its frequency contour. This study explored whether masker impact also depends on spectro-temporal coherence, using a method ensuring that interference occurs only through informational masking. Three-formant analogs of natural sentences were generated using a monotonous periodic source. Target formants were presented monaurally; the target ear was assigned randomly on each trial. A competitor for F2 (F2C) was presented contralaterally; listeners must reject F2C to optimize recognition. In the reference condition, F2C was created by inverting the F2 frequency contour and using a constant RMS-matched amplitude contour. In the coherent conditions, the F2C frequency contour was preserved but the amplitude contour was divided into abutting 100- or 200-ms segments using a raised-cosine envelope (10-ms rise/fall); these values were informed by typical syllable durations. Segment order was randomized in the incoherent conditions, introducing abrupt discontinuities into the F2C frequency contour. Adding F2C lowered keyword scores, but to the same extent for the reference, coherent, and incoherent conditions. This suggests that the impact on intelligibility depends critically on the overall extent of frequency variation in the interferer, but not on its spectro-temporal coherence. [Work supported by ESRC.]

2aSC14. Spectral contrast effects in vowel categorization by listeners with sensorineural hearing loss. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu) and Josh Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

The auditory system is highly sensitive to changes in acoustic input. This is especially true for sounds with relatively stable (reliable) spectral properties across time. Over a limited range, changes to the spectrum (e.g., spectral peak location upon introduction of a new sound) are perceptually enhanced in proportion to the property's long-term reliability, producing spectral contrast effects (SCEs). For example, a neutral vowel between /ɪ/-/ɛ/ is more likely to be labeled /ɛ/ (high-F_1) when preceded by sounds with a reliable low-frequency peak in the F_1 range for /ɪ/, and *vice versa*. Yet, it is unknown how SCEs affect speech perception by hearing-impaired (HI) listeners because research has only examined normal-hearing (NH) listeners. Here, listeners with mild-to-moderate HI identified target vowels varying from /ɪ/-/ɛ/ that followed a precursor sentence. Reliability of precursor spectral peaks was manipulated using low-F_1 or high-F_1 bandpass filters with +5 to +20 dB gain. SCE magnitude was proportional to precursor filter gain (like NH listeners), and surprisingly, to the amount of low-frequency hearing loss. Thus, mechanisms responsible for SCEs are not dependent on healthy hearing, and are magnified by auditory filter broadening associated with sensorineural hearing loss. Implications of these findings for speech perception will be discussed.

2aSC15. Intonation perception problems in advanced age. Amebu Seddoh, Afua Blay, and John Madden (Dept. of Commun. Sci. & Disord., Univ. of North Dakota, 290 Centennial Dr. Stop 8040, Grand Forks, ND 58202, seddoh@und.edu)

This study sought to determine whether intonation processing difficulties for elderly people are attributable to stimulus contextual factors. Participants were 20 old (61–84 years) and 15 young (19–29 years) adult native speakers of English. They were presented auditorily a total of 108 English sentences to decode following evaluation of their hearing abilities. Each sentence conveyed either positive or negative emotional meaning and was accompanied or unaccompanied by contextual information. Both groups of participants demonstrated significantly better outcomes on the decoding of the stimuli presented with contextual information (CS) compared to the decontextualized sentences (DCS). The elderly participants performed poorly relative to the young adults on the perception of positive and negative emotions in the DCS, as well as negative emotions in the CS. On the other hand, their performance on perception of positive emotion in the CS was comparable to that of the young adults. These findings are consistent with data that have shown that intonation meaning is context dependent [e.g., Woodland and Voyer, Metaphor Symbol 26, 227–239 (2011)], and that elderly people have greater amygdala activation for positive than negative stimuli [Mather *et al.*, Psychol. Sci. **15**, 259–263 (2004)]. They suggest that intonation perception problems in old age may be influenced by stimulus contextual information, but part of their origin may lie in neurobiological factors such as age related changes in the brain.

2aSC16. Speech recognition thresholds in reverberation: Effects of age and hearing loss. Nirmal Kumar Srinivasan, Michelle R. Molis, and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, nirmal.srinivasan@va.gov)

While young normally hearing individuals can function in moderate amounts of reverberation with a minimal reduction in speech understanding, older individuals with and without hearing loss are much more susceptible to reverberant distortions of the speech signal (Harris and Reitz, 1985; Danhauer and Johnson, 1991; Helfer, 1992). In this experiment, we evaluated the individual contributions of age and hearing loss to speech reception thresholds (SRT) for reverberant speech (T60: 0, 300, 600, and 1000 ms) in the presence of speech-shaped noise. A one-up one-down adaptive-tracking procedure was used to measure SRT in the better ear of each of 27 listeners. Results indicated, as expected, that older-hearing-impaired listeners had higher SRTs as compared to younger and older normally hearing listeners. However, the variability among listeners was very high. Multiple regression analyses predicting SRT with age and hearing loss as factors indicated that hearing loss was the most significantly contributing factor for all four reverberation times tested. The effects of age and hearing loss on speech recognition in reverberation will be discussed in relation to the data available in the literature. [Work supported by VA RR&D I01RX001020.]

2aSC17. Reliability and time to administer visual analog scaling of intelligibility. Samantha T. Mocarski and Peter J. Watson (Speech-Language-Hearing Sci., Univ. of Minnesota - Twin Cities, 164 Pillsbury Dr., Shevlin 115, Minneapolis, MN 55455, pjwatson@umn.edu)

Measuring speech intelligibility is important in assessing the severity of impaired communication; and is used to demonstrate change overtime, e.g. before and after therapy. Orthographic transcription of speech is considered the clinical-gold standard of assessment, but can be time consuming. Many clinicians opt for quicker, subjective scaling such as Likert-interval, percent-estimate, and more recently visual-analog scaling (VAS). Percentage-estimate and Likert-interval scaling produce lower intra- and inter-judge reliability [Schiavetti (1992)] than transcription. Recently, VAS was used to study intelligibility in persons with dysarthria and found that reliability was good [Stipancic *et al.* (2015)] for mild deficits. We compared VAS to transcription under 4 SNR levels of background noise (−8, −5, −2, and 5). This approach allowed us to compare the reliability of VAS to transcription throughout a range of intelligibility levels. Twenty listeners both transcribed and visually scaled the intelligibility of audio-recorded IEEE sentences of a man and woman under the 4 SNR conditions. Reliability of VAS was as good as transcription at the extremes of intelligibility and less so in the middle levels. Transcription took 3x the time of VAS to administer.

2aSC18. Perceived emotion in clear speech: Effect of simulated hearing loss. Shae D. Morgan, Skyler G. Jennings, and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

Previous research suggests that both young normal-hearing and older hearing-impaired listeners judge clear speech as sounding angry more often than conversational speech. Interestingly, older hearing-impaired listeners were less likely than young normal-hearing listeners to judge sentences as angry in both speaking styles, suggesting that age and/or hearing loss may play a role in judging talkers' emotions. An acoustic cue that helps distinguish angry speech from emotionally neutral speech is increased high-frequency energy, which may be attenuated or rendered inaudible by age-related hearing loss. The present study tests the hypothesis that simulating such a hearing loss will decrease the perception of anger by young normal-hearing listeners. Sentences spoken clearly and conversationally were processed and filtered to simulate the average hearing loss of the older hearing-impaired listeners from a previous study. Young normal-hearing listeners were asked to assign each sentence to one of six categories (happiness, sadness, anger, fear, disgust, and neutral). The judgments will be combined across listeners for each sentence, creating a percentage score for each emotion. The results from judgments of filtered stimuli will be compared with those of unfiltered stimuli as well as with results from young normal-hearing and older hearing-impaired listeners from previous studies.

2aSC19. "That sounds like me" Infants prefer vowels with infant vocal resonances. Linda Polka, Matthew Masapollo (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., SCSD, Montreal, QC H3A 1G1, Canada, linda.polka@mcgill.ca), and Lucie Menard (Dept. of Linguistics, Univ. of PQ at Montreal, Montreal, QC, Canada)

Recent research shows that infants listen preferentially to synthesized vowels that specify an infant source and resonance properties over vowels that simulate a adult female (Masapollo *et al.*, 2015). Infants also preferred vowels with infant resonances over vowels with adult vocal resonances when f0 values were matched (210–240 Hz) across resonance types, suggesting that infant resonance is sufficient to elicit this preference. In this study, we investigate whether infants maintain a preference for infant vocal resonances when f0 values are modulated. In experiment 1, infants listened longer to vowels with infant formants and high f0 values (400–450 Hz) than vowels with adult formants and lower f0 (315–360 Hz). In experiment 2, the same preference emerged when f0 values were reversed; infants listened longer to vowels with infant formants and lower f0 values (315–360 Hz) than to vowels with adult formants and higher f0 (400–450 Hz). These findings show that infant resonance is sufficient to elicit this preference. Given that infants begin to produce vowel-like sounds at 3–4 months, these findings also support the "articulatory filter" hypothesis (Vihman, 1993) which claims that infants are perceptually biased toward speech that resembles their own vocal patterns.

2aSC20. Visual influences on the natural referent vowel bias. Matthew Masapollo, Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca), and Lucie Ménard (Univ. of PQ at Montreal, Montreal, QC, Canada)

Research indicates that perceivers (both adult and infant) are universally biased to attend to vowels with extreme articulatory/acoustic properties (peripheral in F1/F2 vowel space). Yet, the nature of this perceptual phenomenon (i.e., the natural referent vowel [NRV] bias) is not fully understood. The present research investigates whether this bias is attributable to general auditory processes or to phonetic processes that track articulatory information available across modalities. In experiment 1, we examined whether adult perceivers are biased to attend to visual information that specifies extreme vocalic articulations. As predicted by the phonetic account, we found a bias favoring relatively more peripheral vowels when only acoustic or only visual speech information was present. In experiment 2, we investigated how the integration of acoustic and visual speech cues influence the effects documented in experiment 1. When acoustic and visual cues were phonetically congruent, a peripheral vowel bias was observed. In contrast, when acoustic and visual cues were phonetically incongruent, this bias was disrupted. Collectively, these results are compatible with the view that the NRV bias is phonetic in nature—the speech processing system appears to be biased toward extreme vocalic gestures, which may be specified in the optic, as well as in the acoustic, signal.

2aSC21. Analysis of available and listener-received phonetic information in babble-masked consonant identification. Noah H. Silbert and Lina M. Zadeh (Commun. Sci. and Disord., Univ. of Cincinnati, French East 344, 3202 Eden Ave., Cincinnati, OH 45267, silbernh@ucmail.uc.edu)

Speech communication commonly occurs in the presence of multiple, non-target talkers. Previous work has shown that the amount of glimpsed target speech is a good predictor of overall intelligibility of babble-masked speech (Brungart *et al.*, 2006, JASA) and that an automatic speech recognition system trained on glimpses closely approximates listener accuracy as a function of the number of babble talkers (Cooke, 2006, JASA). The present work uses a number of machine learning models to analyze the auditory

information available in babble-masked speech and in available glimpses of babble-masked speech. Regularized Linear Discriminant, Support Vector Machine, and Naive Bayes classifiers were fit to modeled auditory representations of babble-masked CV syllables (with C = t, d, s, z, and V = a). The machine learning models outperformed human listeners substantially (>70% versus 54% accuracy, respectively). Analysis of predicted confusion patterns indicates that the Naive Bayes model most closely approximates human error patterns. The effects of variation in the local and absolute thresholds for glimpse calculation are explored with respect to overall accuracy and error pattern prediction in these machine learning models.

TUESDAY MORNING, 24 MAY 2016 SOLITUDE, 8:00 A.M. TO 11:10 A.M.

Session 2aSP

Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics: Acoustic Array Systems and Signal Processing I

Mingsian R. Bai, Cochair
Power Mechanical Engineering, National Tsing Hua University, No. 101, Section 2, Kuang-Fu Road, Hsinchu 30013, Taiwan

John R. Buck, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

2aSP1. The effect of nearby scatterers on the gain achieved by an acoustic array. R. Lee Culver (ARL, Penn State Univ., PO Box 30, State College, PA 16804, rlc5@psu.edu)

The coherent processing of signals from multiple sensors in an array offers improvements in angular resolution and signal-to-noise ratio. When the array is steered in a particular direction, the signals arriving from that direction are added in phase, and any signals arriving from other directions are not. Array gain (AG) is a measure of how much the signal arriving from the steering direction is amplified relative to signals arriving from all other directions. This talk presents measurements along with supporting theory and simulation showing that as scatterer density increases and AG decreases, random phase shifts in individual sensor signals become larger and occur more often, and signal correlation among the sensor is reduced. The theory and numerical simulation linking scatterer density to the AG are in good agreement with the measurements up to the point where multiple scattering becomes important.

8:25

2aSP2. Differential beamforming with circular microphone arrays. Jacob Benesty (INRS-EMT, Univ. of PQ, Montreal, QC, Canada) and Jingdong Chen (Ctr. of Intelligent Acoust. and Immersive Communications, Northwestern PolyTech. Univ., 127 Youyi West Rd., Xi'an, Shaanxi 710072, China, jingdongchen@ieee.org)

Differential beamforming cannot only achieve high directional gain but also can form frequency-invariant beampatterns. Therefore, it has the great potential to be widely used in voice communications to process broadband speech signals. The performance of a differential beamformer in terms of directivity factor (DF), white noise gain (WNG), and frequency invariance of the beampattern depends on many factors including the array geometry, the number of sensors, the inter-sensor spacing, etc. In the literature, most studies on differential beamforming have been focused on linear microphone arrays and not much efforts have been devoted to other array geometries. This paper is devoted to the study of circular microphone arrays. In comparison with linear arrays whose spatial response is a function of the steering angle, circular microphone arrays can have the same spatial response along many different directions. The focal point of this paper is on differential beamforming with uniform circular microphone arrays. The main topics addressed include: (1) major properties of circular microphone arrays, (2) how to design differential beamformers with different beampatterns, (3) how to design differential beamformers with different orders, and (4) how to achieve a good control of the white noise amplification problem, high directional gains, and frequency-independent responses.

8:45

2aSP3. Estimating the spatial spectra of Gaussian processes with co-prime sensor arrays. John R. Buck and Kaushallya Adhikari (ECE, UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, johnbuck@ieee.org)

A co-prime sensor array (CSA) is a nonuniform line array formed by interleaving two undersampled uniform line arrays. The CSA requires fewer sensors to span the same aperture as a densely sampled uniform line array (ULA), allowing the CSA to match the resolution of the ULA for direction of arrival estimation of narrowband planewaves. However, each CSA subarray suffers from aliasing, or grating lobes, due to the spatial undersampling. Vaidyanathan and Pal (2011) proved that if the subarray undersampling factors are co-prime, the aliasing can be unambiguously resolved by multiplying the spatial spectra of the subarrays. This product spatial spectra is the spatial cross-spectral density between the arrays, and is an estimate of the spatial power spectral density (PSD). In this talk, we extend the classic results of Jenkins and Watts (1968) on the periodogram PSD estimator for Gaussian processes to obtain the product processor's bias for spatially wide-sense stationary processes, and the processor's covariance for spatially white Gaussians. Additionally, we demonstrate that the CSA's product PSD estimate is not necessarily positive definite. Consequently, the CSA product spectrum may fail to detect weak signals in the presence of strong interferers. [Work supported by ONR BRC Program.]

9:05

2aSP4. Time-space analysis, synthesis, and room response modeling of virtual-reality sound fields using microphone and loudspeaker arrays. Mingsian R. Bai and Yi Li (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

This paper proposes a unified framework to analyze and synthesize spatial sound fields. In the analysis phase, an unbaffled 24-element circular microphone array (CMA) is utilized to "encode" the sound field based on plane-wave decomposition, whereas in the synthesis phase, a 32-element rectangular loudspeaker array is employed to "decode" the above-encoded sound field based on pressure matching. Plane-wave decomposition can be implemented in two ways. First, a two-stage algorithm uses the minimum variance distortionless response (MVDR) beamformer to estimate the directions of significant plane-wave components. Next, the amplitude coefficients of plane waves are calculated using the Tikhonov regularization (TIKR) algorithm. Alternatively, a one-stage algorithm that treats the sound field as a great number of plane-wave components uniformly distributed at all angles can be used. In this case, either TIKR or convex (CVX) optimization can be utilized to solve the underdetermined problem for a minimum-norm solution or a minimum-cardinality solution. The same technique is also useful in establishing the response model for a reverberation room with all direct sounds and reflections coded into plane-wave components. In the synthesis phase, pressure matching is achieved by the inverse of the preceding room response model for rendering loudspeakers in the reproduction room.

Contributed Papers

9:25

2aSP5. Subarray compressive beamforming for midfrequency active sonar on a cylindrical array. Jonathan Botts (Appl. Res. in Acoust. LLC, 209 N. Commerce St., Ste. 300, Culpeper, VA 22701-2780, jonathan.botts@ariacoustics.com), Jason E. Summers, and Charles F. Gaumond (Appl. Res. in Acoust. LLC, Washington, DC)

Subarray compressive beamforming is adapted to a cylindrical array for midfrequency active sonar detection of low-Doppler targets in reverberant environments. Low Doppler targets are difficult to detect due to spreading caused by side-lobes and the width of the main lobe. Compressive beamforming and, in particular, subarray compressive beamforming will be shown to drastically mitigate Doppler spreading and interference from static objects. A straightforward application of compressive beamforming to simulated data results in nearly perfect suppression of spread due to sidelobes and good isolation of low Doppler targets from the zero Doppler ridge. Artificially subdividing the array allows for multiple fully resolved sparse estimates of the incident sound field. The set of estimates from the set of subarrays may be combined to form a single sparse estimate, often fully suppressing zero Doppler clutter. Performance of both compressive beamforming and subarray compressive beamforming will be demonstrated using simulated data having realistic Doppler spreading due to multipath and environmental sources of Doppler spreading. Performance will be compared to conventional beamforming methods on the cylindrical array. In addition to practical applications, some theoretical challenges with adaptation to the cylindrical array will be discussed. [Work supported by a NAVSEA Phase I SBIR award.]

9:40

2aSP6. Three-dimensional audio immersion: Development and human perception testing of a large scale, 128 loudspeaker surround sound system. Alexander Kern and Michael Roan (Mech. Eng., Virginia Polytechnic Inst. and State Univ., 111 Randolph Hall, Blacksburg, VA 24061, alkern@vt.edu)

Large-scale loudspeaker arrays for immersive audio are beginning to see increased interest in the research and entertainment industries. There are many algorithms to produce 3D sound fields. Although these algorithms are well developed, there is little published work that quantitatively compares their performance in terms of human perception for various numbers of speakers and speaker arrangements. In this talk, we will present listener perception test results using a 3D, 128 loudspeaker array. This loudspeaker array is part of the CUBE theatre at Virginia Tech. The CUBE is a 5-storey 50 ft. x 50 ft. black-box theatre that uses Dante^{TM} audio-over-ethernet to send signals to the 128 loudspeakers. Combined with a 3D motion tracking system and an immersive video vision system. The CUBE provides a very high-resolution test bed for human perception studies. Using speech intelligibility tests with stimulus phrases and various interferers and noise fields, listener tests will be performed for surround sound algorithms such as Higher Order Ambisonics, Vector Based Amplitude Panning, Wave Field Synthesis, and Higher Order HARPEX (High Angular Resolution Plane Wave Expansion). The Higher Order HARPEX decoder is a novel improvement to First Order HARPEX. Also, we will present technical details of the development of a 16-element soundfield microphone.

9:55–10:10 Break

10:10

2aSP7. Partially correlated source decomposition techniques from a beamforming analysis. Blaine M. Harker, Kent L. Gee, and Tracianne B. Neilsen (Dept. Phys. & Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@gmail.com)

Recent improvements have extended the capabilities of beamforming methods from the uncorrelated monopole assumption to enable reconstructions of extended, partially correlated sources. Resulting source information contains both level and coherence properties that can predict the radiated sound field. In addition, the source cross-spectral matrix, while accurate, can in many cases be reduced into a more convenient or physically meaningful set of basis vectors. For example, a wavepacket is a self-coherent distribution that may be used to create a minimal set of mutually orthogonal basis vectors representing the source levels and coherence properties. In this study, a complex, extended, and partially correlated source is modeled, and the simulated near-field measurements are analyzed with inverse techniques (i.e., beamforming). Possible decompositions are evaluated for their ability to accurately represent the source properties, as well as reliably predict the field levels and coherence properties using the fewest basis terms. Decompositions examined include proper-orthogonal decomposition and spatial decomposition based on coherence lengths. The results point towards an improved interpretation of the source reconstructions and methods for predicting the noise environment from an extended, partially correlated noise source. [Work supported by ONR and USAFRL through ORISE.]

10:25

2aSP8. Azimuth-elevation direction finding using one four-component acoustic vector-sensor spread spatially as a parallelogram array. Yang Song (Universitat Paderborn, Paderborn, North Rhine - Westphalia, Germany) and Kainam T. Wong (Dept. of Electron. & Information Eng., Hong Kong Polytechnic Univ., DE 605, Hung Hom KLN, Hong Kong, ktwong@ieee.org)

An acoustic vector-sensor (a.k.a. a vector hydrophone) consists of three uni-axial velocity-sensors (which are oriented perpendicularly with regard to each other) and one pressure-sensor. Song and Wong have demonstrated how to space these four component-sensors apart in three-dimensional space, in order to extend the overall spatial aperture spanned by them, while improving the accuracy in estimating the azimuth-elevation direction-of-arrival of an acoustic emitter incident from the far field. This paper will focus on a special spatial geometry—where the four component-sensors occupy the four corners of a parallelogram in three-dimensional space.

10:40

2aSP9. Complementary arrays above and below the ocean surface. Paul Hursky (HLS Res. Inc., 12625 High Bluff Dr., Ste. 211, San Diego, CA 92130, paul.hursky@hlsresearch.com)

An experiment was performed off the coast of San Diego to study propagation from an aircraft at altitude to a microphone array above the ocean surface and to hydrophone arrays below the ocean surface. An acoustic hailing device was mounted on a small airplane. An in-air array was formed by mounting individual microphones to the frame of a small boat. Several hydrophone arrays were deployed from the boat. The airplane flew lawn mower patterns over these arrays transmitting broadband waveforms at several altitudes. We will discuss propagation modeling, array calibration and beamforming for this configuration. Doppler is a significant factor due to the high speed of the aircraft and due to the speed of sound in air being five times less than in water. We will assess matched filter gain on the air and air-water paths and discuss phenomena that degrade such gains.

10:55

2aSP10. Improving ray-based blind deconvolution of random shipping sources with short arrays in an ocean waveguide using adaptive beamforming. Juan Yang (Inst. of Acoust.,Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, yang_juanacoustics@163.com), Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea), and Karim G. Sabra (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

This paper introduces an improved ray-based blind deconvolution(RBD) algorithm for sources of opportunity, such as shipping noise, recorded on a short vertical line array (VLA) by using adaptive beamforming. The original RBD algorithm [Sabra *et al*., JASA, 2010, EL42-7] relies on first estimating the unknown phase of the random source using broadband beamforming along a well-resolved ray path to estimate the full channel impulse responses (CIR) between the unknown source and the VLA elements (up to an arbitrary time-shift) as well as recovering the radiated signal by the random source. Hence, RBD performance is limited by the VLA ability to separate adjacent ray path, especially for the case of small aperture. To address this limitation, broadband MVDR is used as a high resolution estimator to improve the RBD performance when using single snapshot recordings and short VLAs. To do so, smoothing techniques are used to increase the rank of cross-spectral data matrix since the ray arrivals emanating from the same random source are naturally correlated in an ocean waveguide when using single snapshot recordings. The improvement of this proposed RBD algorithm will be demonstrated using both numerical simulation and at-sea shipping noise data recorded in shallow water.

Session 2aUW

Underwater Acoustics: Target Physics and Scattering

Brian T. Hefner, Chair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—8:30

Contributed Papers

8:35

2aUW1. Target-depth estimation in active sonar. Alexis F. Mours (Gipsa-Lab, Université Grenoble Alpes, 11 rue des Mathématiques - BP 46, Saint Martin d'Hères Cedex 38402, France), Nicolas Josso (Thales Underwater Systems, Valbonne, France), Jérôme I. Mars, Cornel Ioana (Gipsa-Lab, Université Grenoble Alpes, Saint Martin d'Hères Cedex 38402, France, jerome.mars@gipsa-lab.grenoble-inp.fr), and Yves Doisy (Thales Underwater Systems, Valbonne, France)

This paper presents a new target-depth estimation method that is based on ray back-propagation with a probabilistic approach. This localization algorithm tries to minimize the mean-squared error of elevation angles at the receiver and arrival times between a model and measures. This method was tested on Monte-Carlo simulations of classic active sonar scenarios and using experimental data from a real tank. In active sonar with a point-target model, combined acoustic paths may exist. These have a different path between the sonar-target and the target-array. This paper discusses also about this ray identification. Simulations with vertical array suggest that the target-depth estimation can be realized with a low uncertainty compared to the water column for long ranges in a Mediterranean sound speed profile. However, some environmental parameters as random sound-speed profile, array depth or array tilts could increase the bias and the variance of the target-depth estimator. Results on experimental data with surface noise reveal a good estimation of the target depth and validate our localization algorithm on a constant sound-speed profile with a vertical array.

8:50

2aUW2. Three-dimensional underwater target localization with circular vector sensor array in strong reverberation environment. Ge Yu and Shengchun Piao (College of Underwater Acoust. Eng., Harbin Eng. Univ., Bldg. 145, Nantong St., Nangang District, Harbin 150001, China, liz.221@163.com)

As reverberation can be regarded as an output of time-varying stochastic filtering of the emitted signal, it always leads to high false alarm in shallow water. In this case, some methods have been used to suppress reverberation such as empirical mode decomposition filter, match filter, etc. However, a strong correlation between transmitted signal and reverberation restricts the performance of these methods. In this paper,a novel method based on a circular vector sensor array is proposed to detect and locate multiple targets, respectively, in a strong reverberation environment by using the difference of the Doppler shift distribution between reverberation and target echo. First, spherical harmonic decomposition and beamforming are used to process the wideband signal and generate multiple beams which contain spatial information of interested region. Then, fractional Fourier transform is adopted to get Doppler shift distribution and localization information of reverberation and target echo. Multiple targets can be detected and located separately by estimating variance fluctuation associated with the reflections in different beams. Simulation result shows that this method performs efficiently and reliably in server reverberation environment without any prior information.

9:05

2aUW3. Target strength observations of wobbly bubbles. Alexandra M. Padilla (School of Marine Sci. and Ocean Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, apadilla@ccom.unh.edu), Kevin Rychert, and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Large methane bubbles released from shallow water seabed are of interest because bubbles facilitate the transport process of gas through their journey in the water column, making them more susceptible to reaching the atmosphere than those released from deep water. Several models that relate bubble radius and acoustic backscattering for ideal, spherical bubbles have been developed and are typically used by researchers attempting to invert acoustic backscatter measurements for bubble size and/or gas flux. However, based on theory and field experimentation, it has been shown that large free gas bubbles in liquids (large Eötvös and/or large Reynolds numbers) are non-spherical. A shallow water (<6 m) tank experiment has been designed to understand the influence of bubble shape on acoustic backscattering. This experiment was conducted for bubbles between 1 and 5 mm in radius (Eötvös number between 0.5 and 13, Reynolds numbers between 300 and 1500, and a Morton number of approximately $7.2 \times 10-^{11}$) and at frequencies between 10 and 100 kHz. These frequencies are well above bubble resonance but cover the range of typical echo sounder measurements of bubbles in the field. Calibrated bubble target strength for these bubbles will be compared to existing acoustic backscattering models for bubbles.

9:20

2aUW4. Theoretical and experimental study of a thin elastic cylindrical shell subjected to a point source. Soheil Shah-Hosseini, Fernand Leon, Farid Chati, Dominique Decultot, and Gerard Maze (Laboratoire Ondes et Milieux complexes, UMR CNRS 6294 - Université du Havre, 75, rue Bellot, Le Havre, Haute-Normandie 76600, France, soheil.shahhosseini@gmail.com)

The aim of this work is to study theoretically and experimentally, the acoustic radiation of a cylindrical elastic shell excited by a point source with no internal loading and immersed in a water tank. We are primarily focused on the theoretical study of a cylindrical elastic shell receiving a point force in the form of a Dirac pulse and the acoustic pressure radiated along the tube into a point at far-field observation. The model used is based on the theory of elasticity. It is therefore interesting, in a second step, to conduct experiments to validate these theoretical calculations and to realize complementary measures corresponding to the case of a focalized observer. The used shell is a cylindrical stainless steel tube characterized by its ratio equal to 0.94. Two experiments are carried out. The first experiment is based on a bistatic method, and the transmission and reception are performed by two focalized transducers, characterized by a center frequency of 1 MHz. For the second experiment, the two transducers are different in nature; the transducer acting as the receiver is replaced by a plane transducer characterized by a center frequency identical to the transmitter focalized transducer

9:35

2aUW5. Modes of targets in water and on sand driven by modulated radiation pressure of focused ultrasound. Timothy D. Daniel, Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ivars Kirsteins (NUWC, Newport, RI), and Ahmad T. Abawi (HLS Res., La Jolla, CA)

In a previous work, we had shown that low frequency flexural modes of a circular plate could be excited using modulated radiation pressure generated by focused ultrasound [J. Acoust. Soc. Am. **137**, 2439 (2015)]. Recently, we conducted another experiment where we examined plate, spherical, and cylindrical targets with a larger, higher power focused ultrasound, transducer operated at a carrier of 0.5 or 1.5 MHz that was capable of producing much stronger acoustic radiation pressures. A laser vibrometer was acquired to obtain quantitative measurements of the surface velocity of the targets to facilitate analysis of the radiation force generated on the targets as well as the pressure levels of the target sound emissions. The vibrometer was able to identity modes as well as measure mode shape. Furthermore, we were able to estimate the amount of radiation force that was driving the targets by using the vibrometer measurements to calculate their radiation loading from the water. We were also able to measure the response of targets proud on sand, compared to previous "free-field" experiments, and detect resonant modes and scan the source to infer mode shapes of proud targets with both a hydrophone and vibrometer. [Work supported by ONR.]

9:50

2aUW6. Circular synthetic aperture sonar images of backscattering enhancements on solid elastic cubes due to Rayleigh waves and other mechanisms. Viktor Bollen (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu), Daniel S. Plotnick (Acoust., Appl. Res. Lab., Seattle, WA), and Philip L. Marston (Washington State Univ., Pullman, WA)

Aspect dependent backscattering enhancement mechanisms aid in detecting and identifying underwater objects. This work examines enhancement mechanisms on a solid metal cube observed in acoustic images, emphasizing Rayleigh type surface waves that undergo retroreflection at the corners of the face of the cube [K. Gipson and P. L. Marston, J. Acoust. Soc. Am. **105**, 700–710 (1998)]. Backscattering from a steel cube insonified in water was measured using a circular synthetic aperture sonar system. Image reconstruction was performed using Fourier based algorithms. Rayleigh waves are launched on a face of the cube when the local incident angle is equal to the Rayleigh coupling angle. The presence of four right-angle retroreflectors on each face means that multiple reverberations of this wave are observed. In reconstructed images of the cube, the Rayleigh wave mechanism is responsible for some of the brightest features. If the relative orientation of the circular aperture and the cube is such that specular glints are suppressed, edge and corner diffraction features are clearly observed. This relative orientation may be deduced from the reconstructed image. Discussions of both the spectral response and temporal response of the cube as a function of viewing angle will also be included. [Work supported by ONR.]

10:05–10:20 Break

10:20

2aUW7. Frequency response of proud and partially exposed spheres at flat interfaces. Aaron M. Gunderson and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, aaron.gunderson01@gmail.com)

The backscattering spectrum of spheres at flat interfaces viewed at small grazing angles depends upon reflections from the sphere's surface and the interface, as well as the sphere's elastic response and Franz wave contributions. For partially exposed spheres, one approach to modeling the reflective contribution is to extend the Kirchhoff approximation [K. Baik and P. L. Marston, IEEE J. Ocean. Eng. **33**, 386–396 (2008)] to spheres. Laboratory measurements were carried out using short tone bursts for solid spheres at air-water and sand-water interfaces in such a way that it was typically possible to distinguish between the reflective and delayed elastic contributions viewed in the time domain. For spheres at an air-water interface, the reflective contributions have strong similarities with the Kirchhoff approximation while the strength of the delayed elastic contributions depends strongly on the extent of exposure of elastic guided wave coupling regions. The spectra of proud targets have similarities with image approximations, yielding ridges and valleys in the frequency versus grazing angle domain (associated with interference of contributions of similar magnitude), modulated by elastic contributions. Elastic contributions from small spheres near an air-water interface display a Lloyd's mirror pattern. [Work supported by ONR.]

10:35

2aUW8. Effects of vertical obliquity on sonar images and elastic mechanisms for cylinders near a flat interface. Daniel Plotnick (APL, Univ. of Washington, 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com) and Philip Marston (Washington State Univ., Pullman, WA)

The backscattered signal from targets located near a flat interface is highly dependent on the relative positions and orientations of the acoustic source/receiver, the interface, and the target. The acoustic backscattering spectrum versus aspect angle, also called the "acoustic color'' or "acoustic template,'' of solid cylinders was previously studied for the case where the cylinder axis was vertically oblique relative to a nearby water-air interface [D. Plotnick *et al.*, J. Acoust. Soc. Am. **137**, 470–480 (2015)]. The presence of the interface allows for multiple orientation-dependent paths by which sound is backscattered; this work examines the effects of vertical obliquity on these paths. Experimental data were on a solid cylinder at various obliquities near an air-water interface collected using a circular synthetic aperture sonar system. Emphasis will be placed on the effects of vertical obliquity on features in reconstructed sonar images. Several robust orientation dependent features are considered and the physical mechanisms identified through geometric arguments. Information about a target's 3-D orientation and size may be gleaned from these images, aiding the interpretation of features within the target's acoustic template. The coupling conditions for surface elastic waves are also considered. [Work supported by ONR.]

10:50

2aUW9. A comparison of the predictions of two competing sediment acoustic models for the scattering from inclusions in a sandy sediment. Anthony L. Bonomo and Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

Currently, there are several competing models that have been used to describe the acoustic properties of sandy sediments. These models include those that assume the sediment to behave as an acoustic fluid, a viscoelastic solid, and as a porous medium following Biot theory. Perhaps the two most sophisticated acoustic models that have been applied to sand are the Viscous Grain Shearing (VGS) model of Buckingham and the Extended Biot (EB) model of Chotiros. While both of these models have been used to fit measured sound speed dispersion and attenuation data, the reflection and scattering predictions made using these models differ. In this work, the finite element method is used to compare the scattered pressure predictions of these two models for the case of a sand half-space with a single inclusion. The geometry is assumed axisymmetric, and the inclusions are allowed to be either a fluid or an elastic solid. [Work supported by ONR, Ocean Acoustics.]

11:05

2aUW10. Modeling of sub-bottom volume scattering from turbidite sequences. Derek R. Olson and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, dro131@psu.edu)

Abyssal sediments near the continental margin are often formed through turbidite deposition, which creates alternating layers of silt and mud. Modeling of reflection and sub-bottom volume scattering from turbidite sequences has been previously studied using an effective medium to model the propagation within the sediment, which is only applicable when the acoustic wavelength is much larger than the average layer thickness in the seafloor, and thus is limited to low frequencies. A volume scattering model is developed here to extend the frequency range of previous models of scattering from turbidite sequences. This model employs the Born approximation, takes into account the correct propagation physics in a layered environment, and uses a point source as an incident field. Model results will be compared to previously reported scattering data collected in the Ionian sea.

11:20

2aUW11. Statistics of near surface underwater scattered sound including wave induced Doppler. Sean Walstead (Sensors and Sonar Systems, Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, sean.walstead@navy.mil)

The interaction of underwater sound with the ocean surface is explored. Analytic expressions for the Doppler shift and Doppler spread of surface reflected signals are developed. Statistical models for surface scattered intensity, arrival time, and Doppler fluctuations are developed in relation to wave shape, transducer position, and transmission frequency. Measurements from two experiments are considered. Data collected from AUTEC explore surface scattering effects in the frequency regime of 8 kHz–10 kHz, while data collected in a variable speed wind wave channel at Scripps Institution of Oceanography explore scattering statistics in the 200 kHz–400 kHz (VHF) frequency regime. Physics based analytic expressions for the equations of motion of specular reflection points (SRPs) along a time-evolving wave surface are derived. The time-varying Doppler shift of VHF underwater acoustic signals is attributed to SRP motion associated with surface wave propagation. Model predicted values are compared to experimental measurements and differences between the two tested frequency regimes are highlighted. This work has direct application to the improved performance of phase coherent underwater acoustic communications systems and the remote sensing of gravity-capillary waves.

Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 43/SC 3, Underwater acoustics
ISO/TC 108, Mechanical vibration, shock and condition monitoring,
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 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machine systems, and
IEC/TC 29, Electroacoustics

R.D. Hellweg, Chair and P.D. Schomer, Vice Chair, U.S. Technical Advisory Group for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise
Hellweg Acoustics, 13 Pine Tree Road, Wellesley, MA 02482
Schomer and Associates, 2117 Robert Drive, Champaign, IL 61821

M. A. Bahtiarian, Chair, U.S. Technical Advisory Group for ISO/TC 43/SC 3 Underwater acoustics
Noise Control Engineering, Inc., 799 Middlesex Turnpike, Billerica, MA 01821

W. Madigosky, Chair of the U.S. Technical Advisory Group for ISO/TC 108 Mechanical vibration, shock and condition monitoring
MTECH, 10754 Kinloch Road, Silver Spring, MD 20903

M. L'vov, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures
Siemens Energy, Inc., 5101 Westinghouse Blvd., Charlotte, NC 28273

D. J. Evans, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
13 Watch Hill Place, Gaithersburg, MD 20878

D. D. Reynolds, Chair, U.S. Technical Advisory Group for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Court, Las Vegas, NV 89120

D. J. Vendittis, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machine systems
701 Northeast Harbour Terrace, Boca Raton, FL 33431

D. A. Preves and C. Walber, U.S. Technical Co-advisors for IEC/TC 29, Electroacoustics
Starkey Hearing Technologies, 6600 Washington Ave., S., Eden Prairie, MN 55344 (D. Preves)
PCB Piezotronics, Inc., 3425 Walden Avenue, Depew, NY 14043 2495 (C. Walber)

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting.

The meeting of the Standards Committee Plenary Group will follow the meeting of Accredited Standards Committee S2, which will be held on Monday, 23 May 2016 from 5:00 p.m. - 6:00 p.m.

The Standards Committee Plenary Group meeting will precede the meetings of the Accredited Standards Committees S1, S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:

Tuesday, 24 May 2015	11:00 a.m. - 12:15 p.m.	S12, Noise
Tuesday, 24 May 2016	2:00 p.m. - 3:00 p.m.	ASC S3/SC 1, Animal Bioacoustics
Tuesday, 24 May 2016	3:15 p.m. - 4:30 p.m.	ASC S3, Bioacoustics
Tuesday, 24 May 2016	4:45 p.m. - 5:45 p.m.	ASC S1, Acoustics

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3, and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	**ISO/TC 43/SCI** Noise	ASC S12
M. A. Bahtiarian, Chair	**ISO/TC 43/SC 3**, Underwater acoustics	ASC S1, ASC S3/SC 1 and ASCS12
W. Madigosky, Chair	**ISO/TC 108** Mechanical vibration, shock and condition monitoring	ASC S2
M. L'vov, Chair	**ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures	ASC S2
D. J. Evans, Chair	**ISO/TC 108/SC3** Use and calibration of vibration and shock measuring instruments	ASC S2
D. D. Reynolds, Chair	**ISO/TC 108/SC4** Human exposure to mechanical vibration and shock	ASC S2 and ASC S3
D. J. Vendittis, Chair	**ISO/TC 108/SC5** Condition monitoring and diagnostics of machine systems	ASC S2
IEC		
D. A. Preves and C. Walber U.S. Technical Co-advisors	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

Meeting of Accredited Standards Committee (ASC) S12 Noise

S. J. Lind, Vice Chair, ASC S12
The Trane Co., 3600 Pammel Creek Road, Bldg. 12-1, La Crosse, WI 54601 7599

D. F. Winker, Vice Chair, ASC S12
ETS-Lindgren Acoustic Systems, 1301 Arrow Point Drive, Cedar Park, TX 78613

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.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1, Noise, and ISO/TC 43/SC 3, Underwater acoustics, take note that the meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2016..

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, 24 MAY 2016 SNOWBIRD/BRIGHTON, 1:00 P.M. TO 5:50 P.M.

Session 2pAA

Architectural Acoustics, Noise, and Engineering Acoustics: Predicting and Controlling Heating, Ventilating, and Air Conditioning Systems Noise

Robert C. Coffeen, Cochair
Architecture, University of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045

Andrew N. Miller, Cochair
Bai, LLC, 4006 Speedway, Austin, TX 78758

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. Heating, ventilating, and air conditioning systems noise and electronic sound masking systems in buildings. Andrew N. Miller (BAi, LLC, 4006 Speedway, Austin, TX 78758, amiller@baiaustin.com)

HVAC systems are not typically adequate for both sound masking and neutral background noise quality. Electronic sound masking system cost and overall background noise quality can benefit from considering HVAC system noise character when designing sound masking systems. The contributions of adding electronic sound masking to neutralize background noise character generated by HVAC systems in buildings are explored. Results from general investigations on the topic are presented. General relationships of background noise to speech and noise reduction of construction assemblies are discussed.

1:25

2pAA2. Heating, ventilation, and air conditioning noise calculations after the slide rule and calculator. Steven Thorburn (Thorburn Assoc. Inc., 20880 Baker St., PO Box 20399, Castro Valley, CA 94546, sjt@TA-Inc.com)

Computer programs that predict HVAC system noise are now common. But, what was the basis of the design of the programs, what are the noise control assumptions? This paper will present how the 1984 ASHRAE Chapter 37 Sound and Vibration from the Systems Handbook was converted into a computer program and the limitations that were found in the chapter and how they were modified to more accurately predict the HVAC system noise and it control. This program has been updated 4 times over the years and is now an EXCEL bolt on.

1:45

2pAA3. Sound data for a new format of short-profile fan coil units. K. Anthony Hoover and Brandon W. Cudequest (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

In recent years, a new format of short-profile fan coil units as offered by several manufacturers has become increasingly popular. The sound data for such units are generally plotted in octave bands on NC curves, and have the appearance of great data. However, any scrutiny raises a variety of concerns. Fundamental methods were used to approximate power levels from the provided pressure levels for use in HVAC system noise calculations on various projects. In July 2015, test lab power level data were serendipitously provided for two fan coil units that had been included in the design for a new project. In due course, test lab data for six units, based on AHRI Standard 260, were obtained. This paper will discuss the approximation methods, ranges of differences between pressure level and power level data, effects on results of calculations, referenced measurement standards, and an issue related to pure tones.

2:05

2pAA4. A comparison of three manufacturer's mechanical noise prediction programs and how estimated noise levels from each program compare to field measurements in the subject rooms. Eric McGowan and Robert Coffeen (The Univ. of Kansas, 1465 Jayhawk Blvd., Lawrence, KS 66045, eric.mcgowan789@gmail.com)

The estimation of noise due to heating, ventilation, and air conditioning systems using a program supplied by a manufacturer is a common practice in acoustical consulting. There is little published research comparing these programs against each other and comparing the noise levels predicted by these programs against actual measurements taken in the field. Three rooms on campus at The University of Kansas and one multipurpose room off campus were modeled as similarly as possible in these programs using as-built mechanical drawings and manufacturer's noise data. In addition, ambient noise measurements were taken in each space with the mechanical system operating at a "normal" capacity for springtime in Kansas. The results were used to attempt to determine how the programs compared to each other such as which one was most conservative and which program consistently came closest to predicting the actual noise levels measured in the field.

2:25

2pAA5. Status and dissemination of the current calculation algorithms used in the American Society of Heating and Refrigeration Engineers handbooks. Gregory A. Miller (Threshold Acoust., LLC, 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, gmiller@thresholdacoustics.com)

The Fundamentals and Applications Handbooks by the American Society of Heating and Refrigeration Engineers (ASHRAE) include chapters on noise and vibration control that provide the foundation of most HVAC noise control calculations, whether in commercially available software or in custom software developed by practitioners. The handbooks, however, do not always delve into the detailed calculation algorithms that were used to develop the published tables and charts nor do they provide detailed descriptions of the assumptions and limitations of various calculation methods. ASHRAE last published a compendium of all the calculation algorithms in 1991; the handbooks have been updated to reflect recent research, but without a corresponding update of the detailed algorithms. This paper will summarize ways in which the algorithms have changed since 1991 and discuss the nascent work by the ASHRAE Technical Committee on Sound and Vibration Control (TC 2.6) to compile a new updated compendium of algorithms.

2:45

2pAA6. How audible tones affect psychoacoustic perception of heating, ventilation, and air conditioning noise. Joonhee Lee and Lily M. Wang (Durham School of Architectural Engineering, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, joonhee.lee@huskers.unl.edu)

Assessment of noise from heating, ventilation, and air-conditioning (HVAC) equipment using current building noise rating systems like Noise Criteria or Room Criteria is not reliable when the noise signals include audible tones. Equipment manufacturers and the acoustical consulting community have therefore been seeking new tone criteria for HVAC noise. This paper reviews the current knowledge base, regarding perception of tones and development of tone criteria. Two recent subjective investigations by the authors are presented. The first study utilized multidimensional scaling analysis (MDS) to identify key psychoacoustic parameters to predict human responses to HVAC noise. The second study aimed to develop a more precise method to quantify annoyance perception when multiple tones are present. Results have been used to develop an annoyance prediction model for HVAC noise with tones, which may be used to set evidence-based tone criteria.

3:05

2pAA7. Predicting airborne sound levels from mechanical equipment rooms. Felicia Doggett (Metropolitan Acoust., LLC, 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, f.doggett@metro-acoustics.com)

Predicting airborne sound levels from mechanical equipment rooms or rooftop equipment to adjacent occupied spaces has traditionally been a bit of a gray area when it comes to modeling. Over the last two decades, we have attempted to refine our in-house tools to predict sound levels accurately, but there is much information from various sources and it has been difficult to know what is right and what is not so right. The equation NR = TL + 10log(a/S) only applies to reverberant rooms; other equations from Long, Rathe, among others attempt to refine this equation for other conditions including separating out direct and reverberant sound fields. This presentation explores these various modeling techniques and raises the question as to which one may be "most correct" to use for various conditions and situations.

3:25

2pAA8. Linking design decisions to speech intelligibility. Andrew Hulva, Michael Ermann, Kirsten Hull, Randall Rehfuss, Aaron Kanapesky, and Alex Reardon (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu)

What is the largest classroom that will support unamplified communication? What design attributes and acoustical conditions contribute to speech intelligibility, and how much should each design attribute be weighted when predicting speech transmission index (STI)? This line of inquiry aims to correlate room design and mechanical equipment (HVAC) design to speech intelligibility (STI) in university lecture rooms. It seeks to link the STI of classrooms at Virginia Tech to ambient noise levels, reverberation time, impulse response characteristics, room volume, number of seats, distance to nearest HVAC terminal device, receiver location, and speaker source power. Ambient noise measurements and STI will be mapped on a 1 m x 1 m grid for a subset of the measured rooms to visualize the noise on the listener plane and better estimate the haptic component of signal-to-noise ratio in the spaces. The research also will explore quirks in the STI measurement itself, including differences between the direct and indirect measurement methods, differences permitted for test speech level, and differences from seat to seat within a single room.

3:45–4:00 Break

4:00

2pAA9. Using mechanical noise prediction software to provide suitable solutions. Kevin Butler (Henderson Engineers, Inc., 8345 Lenexa Dr., #300, Lenexa, KS 66214, kevin.butler@hei-eng.com)

Mechanical noise prediction softwares have become a useful tool for consultants to provide noise control recommendations. Two recent projects involving duct breakout will be analyzed to discuss how mechanical noise prediction softwares were used to analyze multiple solutions in order to provide a suitable solution for a medical office building and an open office space.

4:20

2pAA10. Noise reduction of heating, ventilating, and air conditioning systems in previously constructed buildings, primarily focusing on health facilities. Logan D. Pippitt (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, ldpippitt@gmail.com)

Excessive heating, ventilating, and air conditioning system noise is a leading cause of much unwanted noise throughout most commercial buildings, some of which are medical facilities with multiple systems implemented to meet the heating, cooling, and ventilation requirements. Even so, multiple opportunities for correction can be found. Many new buildings are being properly constructed with this problem in mind, though this still leaves many issues in previously constructed buildings and facilities where the noise issues have not been addressed. So what can be done? This paper will explore the possible heating, ventilating, air conditioning, and room acoustic noise issues of a particular nursing home in Olathe, Kansas. The building administrator receives complaints from families, residents, and staff members on multiple noise issues that can be directly attributed to HVAC system noise. Issues such as, but not limited to, exterior noise pollution, supply and return HVAC noise, duct breakout noise, fan coil unit noise, roof top unit source noise, and related room acoustics. In addressing these issues, solutions will aim to meet current ASHRAE acoustical guidelines for medical facilities. [This research was conducted through the School of Architecture, Design and Planning at the University of Kansas.]

4:40

2pAA11. Predicting and controlling heating, ventilating, and air conditioning systems noise: A comparison of computer noise control prediction tools and their potential pitfalls. Nicole Cuff (Acentech, 33 Moulton St., Cambridge, MA 02138, ncuff@acentech.com)

Appropriate background noise levels for mechanical systems are critical for a successful use of any project. There are several computer software packages that acousticians use to evaluate the background noise levels in a space; of course, the acoustician needs to understand all of the inputs into the programs to fully understand the outputs. We will discuss strategies to use these noise prediction tools successfully, including how to evaluate noise sources and translate manufacturer data to input into the programs, how to evaluate various noise paths together for one space, and how to consider sound propagation of different noise sources. We will compare several different and widely available computer noise prediction programs and discuss the positives and challenges of using each one, including the author's assessment of which circumstances lend themselves best to one program versus another.

5:00

2pAA12. Insertion loss provided by several commercially available air handling duct wrap materials that are applied to reduce sheet metal duct breakout noise levels. Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, coffeen@ku.edu)

Noise breaking out of sheet metal HVAC duct is often a significant and disturbing noise source for a building space being served by a supply or return duct or through which the supply or return duct is passing. Reducing breakout noise is often attempted by using heavy loaded vinyl duct wrap applied directly to the duct or applied over glass fiber insulation or foam sheet. Duct wrap insertion loss data were collected using a portion of a sheet metal duct with noise produced by a loudspeaker assembly suspended within the duct. Breakout noise levels were measured on adjacent sides of the duct with and without duct wrap. This approximate insertion loss data were obtained by a University of Kansas architecture student as a portion of work leading to a Master of Arts in Architecture degree.

Contributed Papers

5:20

2pAA13. Simulation of noise propagation through heating ventilation and air conditioning ductwork. David W. Herrin and Kangping Ruan (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, dherrin@engr.uky.edu)

Noise primarily propagates through the airspace or breaks out through the walls of HVAC ductwork. Finite element analysis was used to assess the noise attenuation of each of these paths. Specifically, the insertion loss and breakout transmission loss were assessed for both bare and lined ductwork. Sound absorptive lining was modeled using poroelastic finite elements and the ductwork itself was modeled using shell and beam elements. A diffuse field was approximated at the source using a collection of monopole sources having random phase. Results were compared to measurement with good agreement. Several conclusions can be made regarding noise transmission in large ducts that should be transferable to other industries.

5:35

2pAA14. Noise analysis of heating, ventilating, and air-conditioning systems: Modeling in the Trane acoustics program. Jennifer E. Russell (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, jrussell@siebeinacoustic.com)

Many different acoustical issues can arise in a building as a result of heating, ventilating, and air-conditioning (HVAC) systems. These issues can include excessive duct-borne noise, inadequate sound isolation, loud mechanical equipment, and sound paths between spaces created by connected ductwork. In each case, the Trane Acoustics Program (TAP) software can be used to model sound paths and assess the acoustical impact the HVAC system has on a space. This paper will present several case studies representing common mechanical and architectural noise control issues resulting from HVAC system design that were assessed using the TAPsoftware.

TUESDAY AFTERNOON, 24 MAY 2016 — SALON I, 1:00 P.M. TO 3:00 P.M.

Session 2pABa

Animal Bioacoustics and Underwater Acoustics: Cetacean Bioacoustics

Julia Vernon, Chair
Graduate Program in Acoustics, The Pennsylvania State University, State College, PA 16803

Contributed Papers

1:00

2pABa1. What do acoustic tags placed on the back of echolocating dolphins really measure? Whitlow Au (Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu), James J. Finneran (Space and Naval Warfare Systems Ctr., San Diego, CA), and Brian K. Branstetter (National Marine Mammal Foundation., San Diego, CA)

Suction cup deployed acoustic tags have been used to study the echolocation behavior of a number of odontocetes, yet the relationship between the projected biosonar signals and tag recordings are not known. Acoustic data obtained from these tags consist of the number of clicks emitted, the depth at which the clicks are emitted and the inter-click intervals during a biosonar search. In order to understand the relationship between the emitted signals detected in the front of a dolphin and the signals detected by a tag, a spherical hydrophone was mounted on a wooden model of a tag and mounted on the back of a dolphin via a suction cup. The dolphin was involved in a biosonar discrimination task and the signals were measured 1 m from its blowhole along the acoustic axis of the beam and the acoustic tag placed in five different locations on the animal's back. The signals recorded by the tag were complex with the first pulse being a high frequency resonance-like signal followed by clicks resembling the outgoing clicks but experiencing reflective interference. The peak-to-peak amplitude of the signals measured by the tag was between 40 and 50 dB lower than that of the outgoing signal.

1:15

2pABa2. Does depth matter? Investigating the effect of recording depth on delphinid whistle characteristics and classifier performance. Julie N. Oswald (Bio-Waves, Inc., 364 2nd St., Ste. #3, Encinitas, CA 92024, julie.oswald@bio-waves.net), Marc O. Lammers, Anke Kügler (Oceanwide Sci. Inst., Honolulu, HI), Cory Hom-Weaver, and Robyn Walker (Bio-Waves, Inc, Encinitas, CA)

Seafloor acoustic recorders are commonly used to obtain information about cetaceans, as they allow data to be collected for long periods without

the presence of a human operator. Recordings collected using seafloor instruments do not have associated visual observations, so species must be identified based on their calls. Visually validated acoustic recordings are necessary for training acoustic species classifiers and so most are trained using data collected near the sea surface. The suitability of using classifiers trained using surface recordings to analyze recordings obtained at depth is unknown. To investigate this, we used a vertical array of four Ecological Acoustic Recorders (EARs) spaced 90 m apart to record delphinids at different depths. The same whistles were measured from each EAR and median values of 17 spectrographic variables were compared among EARs for six acoustic encounters. For five of the encounters, there were significant differences in whistle variables among EARs, most commonly in frequency variables. When a random forest classifier was used to identify these whistles to species, the same five encounters were classified as different species when recorded at different depths. These results suggest that caution should be taken when applying classifiers developed using surface data to whistles recorded at depth.

1:30

2pABa3. There must be mucus: Using a lumped-parameter model to simulate the "thump" and "ring" of a bottlenose dolphin echolocation click. Lester Thode (Los Alamos, NM), Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), and Whitlow Au (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Kaneohe, HI)

Bottlenose dolphin echolocation clicks display a great diversity in temporal and spectral structure, with both unimodal and bimodal spectra observed (Houser *et al*., JASA, 1999). Wavelet scalograms applied to data collected by the Navy Marine Mammal Program and the Bioacoustic Measuring Tool (BMT) (Martin *et al*., JASA, 2005) show that echolocation clicks can display two distinct phases: an initial "thump," followed by an extended "ring" that is adequately modeled by a damped harmonic oscillator. The thump and ring can display either similar or different spectral characteristics, giving rise to a unimodal or bimodal spectrum. A three-mass lumped parameter model, adapted from the speech processing and terrestrial bioacoustics literature, has been used to simulate the oscillation and collision of the dorsal bursae in a dolphin's nasal passage. The three-mass model reproduces many of the time and frequency domain features of entire click trains as well as individual clicks, including unimodal and bimodal spectra. A key insight of the models is that some slight adhesion between the faces of the colliding bursue seems necessary in order to reproduce the high-frequency click structure. A viscoelastic mucus coating could provide one possible mechanism for this required adhesion force. [Data provided by Steve Martin, NMMF.]

1:45

2pABa4. Inter and intra specific variation in echolocation signals among odontocete species in Hawaii, the northwest Atlantic and the temperate Pacific. Tina M. Yack, Kerry Dunleavy, and Julie N. Oswald (Bio-Waves, Inc., 364 2nd St., Ste. #3, Encinitas, CA 92024, tina.yack@bio-waves.net)

Odontocete species use echolocation signals (clicks) to forage and navigate. The aim of this study is to explore inter- and intra-specific variation in clicks among odontocete species in the Northwest Atlantic, Temperate Pacific, and Hawaii. Clicks were examined for seven species of delphinids in the Northwest Atlantic; common dolphin, Risso's dolphin, pilot whale, rough-toothed dolphin, striped dolphin, Atlantic spotted dolphin, and bottlenose dolphin. Newly developed PAMGuard tools were used to automatically measure a suite of click parameters. Five parameters were compared among species; duration, center frequency, peak frequency, sweep rate, and number of zero crossings. Significant differences in duration, center and peak frequency were evident among species within this study area (Dunn's test with Bonferroni adjustment $p < 0.05$). Geographic variation in click parameters among the three study regions was compared for five species; bottlenose dolphin, common dolphin, striped dolphin, pilot whale, and Cuvier's beaked whale. Significant differences in several parameters were found for all species among the regions (Dunn's test with Bonferroni adjustment $p < 0.05$). These results suggest that there are species specific differences in clicks among delphinids and that geographic variation exists for multiple species. The ecological significance of these findings will be discussed along with implications for classifier development.

2:00

2pABa5. Relative abundance of sound scattering organisms in the Northwestern Hawaiian Islands is a driver for some odontocete foragers. Adrienne M. Copeland (Univ. of Hawaii at Manoa, P.O. Box 1106, Kailua, HI 96734, acopelan@hawaii.edu), Whitlow W. Au (Hawaii Inst. of Marine Biology, Kailua, HI), Amanda Bradford, Erin Oleson, and Jeffrey Polovina (Pacific Islands Fisheries Sci. Ctr., NOAA, Honolulu, HI)

Previous studies in the Northwestern Hawaiian Islands (NWHI) focused on shallower communities in and near reefs and did not investigate the organisms living in deeper waters that some apex predators rely on for food, e.g., some odontocetes forage at depths greater than 400 m. To examine the relationship between deep-diving odontocete predators and prey, a Simrad EK60 echosounder operating at 70 kHz collected acoustic abundance throughout the NWHI from May 7 to June 4, 2013. Visual and passive acoustic surveys for marine mammal presence were conducted concurrently with the echosounder. Two broad scattering layers were found, a deep layer from 325 to 670 m and a shallow layer from 0 to 195 m. The highest densities of both deep and shallow scattering organisms were associated with deep slopes of banks and atolls. Beaked and short-finned pilot whale sightings occurred in locations of high scattering density associated with slopes of atolls and banks. It is hypothesized that the high scattering organisms associated with these features are similar to the mesopelagic boundary community found in the Main Hawaiian Islands and support a food web representing the prey of the cetaceans.

2:15

2pABa6. Beaked whale acoustic versus visual detection. Odile Gerard (DGA Naval Systems, Ave. de la Tour Royale, Toulon 83000, France, odigea@gmail.com)

Because of their sensitivity to anthropogenic noise, research on beaked whale habitat is particularly important. During 2010 and 2011, NATO Undersea Research Centre (NURC) conducted sea trials dedicated to marine mammals, in areas of potential beaked whale habitat. The first one took place in North Eastern Atlantic Ocean, Southwest of Portugal, and the second one took place in the Gulf of Genoa, Mediterranean Sea. For both trials: weather conditions allowing, during daylight there were two teams of visual observers, working in two shifts, scanning the horizon and taking note of marine mammal encounters. Acoustic data were collected with the CPAM (Compact Passive Acoustic Monitoring), designed by NURC. The total usable bandwidth is up to 80 kHz. The CPAM was deployed at a depth between 100 and 200 m for about 20 h a day Beaked whale detection obtained by visual observers and by passive acoustic are analyzed. The number of detections and the information obtained by each method are compared. The advantages and drawbacks are highlighted.

2:30

2pABa7. Seasonal variability in distribution of fin whales around Wake Island. Julia A. Vernon and Jennifer L. Miksis-Olds (Graduate Program in Acoust., Appl. Res. Lab, The Penn State Univ., State College, PA 16803, jav232@psu.edu)

Passive acoustic monitoring in population density estimation of marine mammals provides an efficient and cost-effective alternative to visual surveys. However, one challenge that arises with this method is uncertainty in the animal distribution. Information about distribution is needed in order to account for spatial variability in the probability of detection. Consideration also needs to be given as to how distribution varies between seasons, as seasonal variability also needs to be incorporated into the density estimation. This paper presents bearing estimates of fin whales around Wake Island in the Equatorial Pacific Ocean, using low-frequency ambient noise data (5–115 Hz) acquired by the Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO) International Monitoring System. Bearings were initially calculated using time delay information from the cross-correlation of received signals. However, a simple cross-correlation is not a viable option for many calls, due to distortion of the waveform as a result of modal dispersion, and alternate methods of determining time delays of received signals are discussed. Bearings were calculated for individuals detected over a period of three years: May 2007 to May 2010. Seasonal variability in distribution is presented. [This work was supported by the Office of Naval Research.]

2:45

2pABa8. Long-term monitoring of Physeteroidea (sperm whales, dwarf, and pygmy sperm whales) in the Central and Western Pacific. Karlina Merkens (CRP, NOAA/PIFCS (Lynker Tech.), 3710 SW Caldew St., Portland, OR 97219, kmerkens@gmail.com), Anne Simonis (UCSD/SIO, La Jolla, CA), and Erin Oleson (CRP, NOAA/PIFSC, Honolulu, Hawaii)

The superfamily Physeteroidea includes three extant species: the sperm whale (*Physeter macrocephalus*), the dwarf sperm whale (*Kogia sima*), and the pygmy sperm whale (*K. breviceps*). Despite extreme difference in size between the *Kogia* spp. and their large *Physeter* relative, all three share ecological and acoustic traits relating to their deep-diving behavior and high rates of acoustic activity. All three species can be found across the Central and Western Pacific ocean, an area that has been monitored using passive acoustics (High-frequency Acoustic Recording Packages, HARPs) for more than 10 years. We identified sperm whale and *Kogia* spp. signals in the long-term HARP records from 13 locations across the Central and Western Pacific ocean. A combination of automated tools and human analysis were used to record detection events of both types of signals. While sperm whales were found at all 13 locations, the *Kogia* species (which cannot yet be distinguished acoustically) were detected at approximately half of the sites. Presence of sperm whale signals was modeled to determine if temporal parameters, such as lunar cycle and day of the year, could explain patterns of presence. Across the whole region the best model included the day of the year and the recording site, while sub-regions and site-specific models had slightly different combinations of parameters.

TUESDAY AFTERNOON, 24 MAY 2016 SALON I, 3:30 P.M. TO 5:00 P.M.

Session 2pABb

Animal Bioacoustics: Animal Bioacoustics Poster Session

Benjamin N. Taft, Chair
Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405

All posters will be on display and all authors will be at their posters from 3:30 p.m. to 5:00 p.m.

Contributed Papers

2pABb1. The spatial unmasking of pure tones by laboratory mice. Laurel A. Screven (Psych., Univ. at Buffalo, B29 Park Hall, Buffalo, NY 14260, laurelsc@buffalo.edu)

The ability of animals to identify signals in their acoustical environment when confronted with sounds from multiple sources relies on them extracting the important information and filtering out background noise. Previous reports have shown that many animals are able to attend to a single sound more easily when that signal is spatially separated from the background noise. This spatial unmasking of a sound has previously not been behaviorally measured in mice, despite their use as a model for human hearing and communication. The present experiment examined if laboratory mice are able to show spatial release from masking, as seen in other animals, by testing the detection of 2, 4, 8, 12, 16, and 24 kHz pure tones in the presence of a white noise masker. The masker was spatially coincident with the signal or separated by 90°. We hypothesized that the mice would experience more spatial unmasking for the higher frequency tones since they rely heavily on these frequency ranges for their communication signals. Preliminarily, we have found that mice are able experience spatial unmasking at some, but not at all frequencies.

2pABb2. Intensity difference limens in mice. Anastasiya Kobrina, Katrina Toal, Kali Burke, and Micheal L. Dent (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

The ability to distinguish between sounds of differing intensities is a universal auditory process. Previous researchers have examined the intensity discrimination abilities of goldfish, parakeets, and several mammals including feral and house mice using behavioral approaches. CBA/CaJ mice are frequently used as a model for human hearing research, yet intensity difference limens (IDLs) have not been measured in this strain. The aim of the present study was to establish IDLs for 12, 16, 24, and 42 kHz tones at 10 and 30 dB SL. Eight adult animals were trained and tested in a discrimination task using operant conditioning procedures with positive reinforcement. IDLs were variable across all frequencies for 10 dB SL, but remained stable for 12, 16, and 42 kHz at 30 dB SL, with elevated IDLs at 24 kHz. Calculations of the Weber fractions at each sensation level tested showed that Weber's law does not hold true for IDLs in CBA/CaJ mice. Overall, thresholds were comparable to those found in other animals.

2pABb3. Utilizing the sound of predators to cast out wild animals. Myungsook Kim (English, Soongsil Univ., Sangdo-ro 369, Seoul 06978, South Korea, kimm@ssu.ac.kr), Ik-Soo Ahn, and Myung-Jin Bae (Information and TeleCommun., Soongsil Univ., Seoul, South Korea)

Recently, there have been many reports of damages caused by animals such as wild pigs and wild dogs in the rural farms of Korea. Damages were extensively widespread from agricultural crop and produce to domestic animals. To efficiently cast out wild animals, the recorded sound of predators such as lions and tigers was used. The speakers were installed around the farms and broadcasted the sound of predators every 3 h. Although the damages were a little bit reduced after broadcasting, it did not complete its purpose and still some damages were reported. We analyzed the recording quality of the sound and found out some deficiencies in it, the lack of low frequency sound below 100 Hz in particular. In order to improve its sound quality, we installed the speakers inside of water pipes with 1 m in diameter. As a result, we earned 90% in the index of similarity between the original crying sound and the broadcasted crying sound of predators in its sound spectra. Since the improved sound recording was broadcasted for 1 min every 3 h in the farms, the damages have not been reported.

2pABb4. Acoustical and vibratory characteristics of a synthetic mallard syrinx replica. Taylor B. Groom (Mech. Eng., Brigham Young Univ. Idaho, Rexburg, ID 83460, gro11017@byui.edu), Tobias Riede (Physiol., Midwestern Univ., Glendale, AZ), and Scott L. Thomson (Mech. Eng., Brigham Young Univ. Idaho, Rexburg, ID)

The study of the physics of human voice production has benefited by the use of synthetic human vocal fold replicas. Extending this concept to avian vocalization, a functioning model of a male mallard syrinx for studying the relationship between syrinx anatomical features and vocalization characteristics is described here. The mallard syrinx is characterized by two sets of labia located at the tracheobronchial juncture, and vocalization is produced by the flow-induced vibration of these labia. Shape and histological composition of labia demonstrate similarities to mammalian vocal folds. In this study, life-sized synthetic replicas based on CT data of an adult male mallard syrinx, including airway and labia regions, are studied. The replicas are fabricated using exceedingly flexible silicone materials that have been previously used in synthetic human vocal fold replicas. The replicas are mounted in tube-like airways representing the trachea and bronchi, with flow generated by an adjustable air supply. In this presentation, the CT-based syrinx model, including fabrication process, will be introduced. The model response, including acoustical output and vibratory motion, will be described and compared to previous models and to existing data about mallard vocalization.

2pABb5. Neyman-Pearson detection of underwater bioacoustic signals. Florian Dadouchi (Gipsa-Lab, Univ. Grenoble Alpes, 11 rue des Mathématiques, Grenoble Campus, SAINT MARTIN D'HERES BP46, F-38402 CEDEX, France, florian.dadouchi@gipsa-lab.fr), Julien Huillery (Méthodes pour l'ingénierie des systèmes, Laboratoire Ampère (CNRS UMR5005)., Ecully, France), Cédric Gervaise (Chorus Foundation., Grenoble Cedex 1, France), and Jérôme I. Mars (Gipsa-Lab, Univ. Grenoble Alpes, Saint Martin d'Hères, France)

The use of passive acoustic for the classification, localization, and density estimation of populations of marine mammals is a current area of interest. It is a cheap and an efficient alternative to visual surveys. However, the lack of an efficient automatic detector for unknown marine mammal calls greatly undermines the feasibility of those tasks, especially when dealing with species showing a great variability of calls. This study adds one more step toward the fully automatic detection of unknown bioacoustic signals in impulsive, non-stationary, and colored ocean noise. The detection procedure is a two-steps fully statistical method solely based on the knowledge of the background noise in the spectrogram. The first step models the noise power as a chi-squared distribution, which parameter is estimated. The signal is then detected using a Neyman-Pearson approach, providing a binary spectrogram that contains false and true detections. The second step removes a significant amount of false detections from the binary spectrogram. The time-frequency distribution of false detections is fitted with a correlated binomial distribution, which is used to discriminate patches of detections (signal) from uniformly distributed detections (false alarms). Examples showing the applicability of this method on several real underwater sounds are presented.

Session 2pBA

Biomedical Acoustics: Biomedical Acoustics Student Paper Competition (Poster Session)

Kevin J. Haworth, Chair
University of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD $500 for first prize, USD $300 for second prize, and USD $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee. Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions associated with the abstract numbers.

All entries will be on display and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

2aBAa1. Toward validation of shear wave elastography using vibration rheometry in soft gels
Student author: Sanjay Yengul

2aBAb1. Jet formation of contrast microbubbles in the vicinity of a vessel wall
Student author: Nima Mobadersany

2aBAb4. Tracking kidney stones during shock wave lithotripsy
Student author: Kya Shoar

2aBAb5. Effects of acoustic parameters on nanodroplet vaporization
Student author: Krishna Kumar

2aBAb6. Effects of ultrasound in presence of microbubbles on cartilage tissue regeneration in three-dimensional printed scaffolds
Student author: Mitra Aliabouzar

2aBAb7. Microbubble response to dual frequency excitation for broadband contrast imaging
Student author: Christina Keravnou

2aBAb8. Lytic efficacy of tissue plasminogen activator and ultrasound in porcine clots doped with barium sulfate in vitro
Student author: Shenwen Huang

3aBA7. Ultrasound-mediated transport of nanoparticles and the influence of particle density
Student author: Harriet Lea-Banks

3pBA6. Phased array techniques for multiple focus synthesis in transcranial focused ultrasound
Student author: Alec Hughes

4pBA1. Simulation and implication of a two-dimensional phased array flexible ultrasound system for tissue characterization
Student author: Zhihua Gan

4pBA3. Development of a nonlinear model for the pressure dependent attenuation and sound speed in a bubbly liquid and its experimental validation
Student author: Amin Jafari Sojahrood

4pBA5. Photoacoustic imaging of muscle oxygenation during exercise
Student author: Clayton Baker

Session 2pEA

Engineering Acoustics: General Topics in Engineering Acoustics II

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Chair's Introduction—1:10

Contributed Papers

1:15

2pEA1. Sensitivity analysis of the performance of a linear alternator at mechanical resonance under thermoacoustic power conversion conditions. A. H. Ibrahim (American Univ. in Cairo, Egypt 11835, Egypt, abdelmaged@aucegypt.edu), Ahmed Yassin, and Ehab Abdel-Rahman (American Univ. in Cairo, New Cairo, Egypt)

A linear alternator lie at the interface between the acoustic power generated by a thermoacoustic engine and an electric load, and thus, its performance is significantly controlled by its matching with the thermoacoustic engine and with the electric load. Under thermoacoustic power conversion conditions, small unavoidable changes in the operating conditions may incur significant effects on the alternator performance. In this work, the sensitivities of several alternator performance indices, namely, the acoustic-to-electric conversion efficiency, the mechanical stroke, and some of the main alternator losses (mechanical damping loss, seal loss, and electric copper loss), are examined to small changes in four operating conditions. Using the methodologies of the design of experiments and sensitivity analysis, a scheme of experiments is designed and carried-out to analyze the sensitivity of these indices to $\pm$ 10% changes in operating conditions at mechanical resonance. The operating conditions considered are the gas mixture composition, the mean gas pressure, the dynamic pressure ratio acting on the alternator and the electric load. The results reveal how variations in each of these operating variables as well as variations in their combined interactions affect the alternator's performance with respect to the results obtained in a reference experiment.

1:30

2pEA2. Performance of the phase and amplitude gradient estimator intensity method in interference fields. Darren K. Torrie, Eric B. Whiting, Michael T. Rose, Kent L. Gee, and Scott D. Sommerfeldt (Phys., Brigham Young Univ., Eyring Sci. Ctr., Provo, UT 84606, darren@torriefamily.org)

The phase and gradient amplitude estimator (PAGE) method improves the frequency bandwidth of estimated acoustic intensity over the traditional p-p method without altering the spacing between microphones [D. C. Thomas *et al*., J. Acoust. Soc. Am. **137**, 3366–3376 (2015)]. For many broadband sources, accurate estimates may be obtained beyond the spatial Nyquist frequency by unwrapping the phase of a transfer function used in obtaining the phase gradient. However, inaccurate phase unwrapping in interference fields, such as those produced by two loudspeakers with equal strengths but opposite phase, has been observed. This results in erroneous intensity vectors. A two-dimensional, multi-microphone intensity probe was employed to investigate this phenomenon. Findings include: (a) the unwrapping error does not occur for all interference nulls, but is more likely to occur for deeper interference nulls where there is reduction in coherence; (b) rotation of the probe in the field alters which pair-wise transfer function unwraps erroneously, but does not significantly alter the direction that contains the inaccurate intensity vectors; (c) removal of the interference nulls by driving the loudspeakers as incoherent sources allows for phase unwrapping to occur multiple times and accurate estimation of intensity vectors beyond 10 kHz. [Work supported by NSF.]

1:45

2pEA3. Comparison of pressure-based intensity measurements for different probe designs and estimation methods. Michael T. Rose, Darren K. Torrie, Reese D. Rasband, Kent L. Gee, and Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., Provo, UT 84602, rosemission@surewest.net)

Acoustic intensity measurements have traditionally used cross spectral methods with multimicrophone probes to estimate the required pressure and particle velocity. The phase and gradient estimator (PAGE) method increases probe bandwidth without modifying microphone spacing as compared to the traditional cross spectral method [D. C. Thomas *et al*., J. Acoust. Soc. Am. **137**, 3366–3376 (2015)]. In this study, high-frequency probe bandwidth obtained using both the PAGE method and the traditional method is compared across three different multimicrophone probe designs: a three-axis, six-microphone probe with variable solid spacer, a three-axis, four-microphone spherical probe, and a two-axis, four-microphone probe developed in-house for rocket measurements. Broadband, anechoic loudspeaker measurements are used to examine the performance of each probe and estimation method. These include a single 7.6 cm diameter, enclosed loudspeaker driver, and two drivers of equal strength but opposite phase. Intensity measurements in a grid of 41 $\times$ 41 points inside a one square meter area are presented. Probe design robustness is determined by considering bias errors in calculation methods and probe scattering. [Work supported by NSF.]

2:00

2pEA4. Adaptive imaging processing for reducing ultrasonic ringdown interference pattern. Zhijuan Zhang, Douglas Patterson, Roger Steinsiek, and Wei Han (Baker Hughes, 2001 Rankin Rd., Houston, TX 77073, Zhijuan.Zhang@bakerhughes.com)

In the oil and gas industry, ultrasonic imaging tools are actively used in stress identification and fracture detection. Pulse-echo transducers measure the properties of the echoes reflected from the formation, and the signal peak amplitude and travel time are typically displayed as a circumferential image of the borehole wall. A well-known problem is that the pulse-echo transducers have internal ringdown which constantly modulates formation echoes. As a result, the acquired image is often dominated by a "wood-grain" interference pattern which degrades image quality. Traditional ringdown reduction methods try to remove the ringdown effect by estimating the ringdown waveform. However, the ringdown signal varies versus borehole conditions, which make it inadequate for traditional methods to efficiently reduce the image interference pattern. In this paper, we proposed an image processing method which significantly reduces the interference pattern. This new image processing method does not require any ringdown

waveforms. It adaptively estimates the ringdown modulation template and after that constructs the interference amplitude pattern. Examples shows promising results and have proven the propose method is more efficient than traditional ringdown reduction methods on the ringdown interference pattern reduction.

2:15

2pEA5. *In-situ* ultrasonic measurements of creep specimens. Manton J. Guers (Structural Acoust., Penn State Appl. Res. Lab., PO Box 30, State College, PA 16804, mjg244@psu.edu) and Bernhard R. Tittmann (Eng. Sci. & Mech., Penn State Univ., University Park, PA)

Performing *in-situ* measurements of specimens in research reactors is challenging because of the environmental conditions. In this work, ultrasonic guided waves were investigated for performing *in-situ* measurements of the change in length of creep specimens. Both theoretical calculations and experimental measurements were used to determine the proposed method's sensitivity to changes in temperature and elongation. The experimental tests demonstrated that careful consideration must be given to the signal processing of the data. Successful measurements of the creep elongation of a 3 in. gauge length specimen were demonstrated.

2:30–2:45 Break

2:45

2pEA6. Active compensation of nonlinear distortion in balanced armature receivers. Buye Xu and Tao Zhang (Signal Processing Res., Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com)

Balanced armature receivers have been widely used in hearing aids due to their higher efficiency and more discrete size comparing to moving coil loudspeakers. For the application of hearing aids, it may require a very high output sound pressure level (up to 110 dB in an opened ear canal) and, yet, minimum nonlinear distortions. This requirement often results in a larger receiver in size, which is less comfortable for wearing. It is highly desired if the linear range of a smaller receiver can be broaden without increasing its physical size. Active compensation of nonlinearity for moving coil loudspeakers through signal processing means have been well studied. However, similar research for balanced armature receivers is rarely seen in the literature. Although the two types of transducers differ in many aspects, they share commonalities in the nonlinearity mechanism. The current study surveys the active compensation techniques developed for moving coil loudspeakers and explores the possibility of adopting them to reduce the nonlinearity of balanced armature receivers.

3:00

2pEA7. Size differentiation of a continuous stream of particles using acoustic emissions. Ejay Nsugbe, Andrew Starr, Peter Foote, Cristobal Ruiz-Carcel, and Ian K. Jennions (Cranfield Univ., 79 Lower Shelton Rd., Marston Moretaine MK43 0LN, United Kingdom, e.nsugbe@cranfield.ac.uk)

Procter and Gamble (P&G) requires an online system that can monitor the particle size distribution of their washing powder mixing process. This would enable the process to take a closed loop form which would enable process optimization to take place in real time. Acoustic emission (AE) was selected as the sensing method due to its non-invasive nature and primary sensitivity to frequencies which particle events emanate. This work details the results of the first experiment carried out in this research project. The first experiment involved the use of AE to distinguish sieved particle which ranged from 53 to 250 microns and were dispensed on a target plate using a funnel. By conducting a threshold analysis of the peaks in the signal, the sizes of the particles could be distinguished and a signal feature was found which could be directly linked to the sizes of the particles.

3:15

2pEA8. Recording anechoic gunshot waveforms of several firearms at 500 kilohertz sampling rate. Tushar K. Routh and Robert C. MAHER (Dept. of Elec. and Comput. Eng., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717, tushar.routh@msu.montana.edu)

Acoustic gunshot signals consist of a high amplitude and short duration impulsive sound known as the muzzle blast. This experiment involved documenting gunshot muzzle blast sounds produced by eight commonly used firearms. An elevated bracket (3 m above the ground) was built to achieve a quasi-anechoic environment for the duration of the muzzle blast. Twelve microphones (GRAS 46DP) were mounted on the bracket in a semi-circular arc to observe the azimuthal variation of the muzzle blast. Signals were recorded using LabVIEW. Similarities and differences among waveforms are presented.

3:30

2pEA9. Vorticity-involved acoustic damping performance of an in-duct orifice with a bias flow. Dan Zhao, Nuomin Han (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore 639798, Singapore, zhaodan@ntu.edu.sg), and Xiao Jin (School of Energy and Power Eng., Jiangsu Univ. of Sci. and Technol., Zhenjiang City, Jiangsu, China)

Acoustic damping performance of an in-duct perforated orifice with a bias flow in terms of power absorption and reflection coefficients are evaluated in this work. For this, experimental measurements of a cold-flow pipe system with a diameter of $2b$ with an in-duct perforated plate implemented are conducted first. It is shown that the maximum power absorption Δ_{max} and reflection coefficients χ_{max} are approximately 80% and 90%, respectively. In addition, Δ and χ are periodically changed with the forcing frequency. To simulate the experiments and gain insights on the damping performance of the orifice with a diameter of $2a$, a 1D theoretical model embodying vorticity-involved damping mechanism is developed. It is based on the modified form of the Cummings equation describing unsteady flow through an orifice and the Cargill equation describing acoustically open boundary condition at the end of the downstream duct. It is shown that Δ and χ are strongly related to (1) the mean flow Mach number, (2) forcing frequency ω, and (3) porosity $\eta = a/b$, and (4) the downstream pipe length L_d. Theoretical predictions are found to agree well with experimental measurements. This confirms that the model has the potential to predict the acoustic damping performance of in-duct orifices.

3:45

2pEA10. Absorption of axial plane waves by using double- and single-layer perforated liners. Dan Zhao, Nuomin Han (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore 639798, Singapore, zhaodan@ntu.edu.sg), and Linus Ang (Dept. of Mech. Eng., National Univ. of Singapore, Singapore, Singapore)

In this work, the acoustic damping performances of multiple double- and single-layer perforated liners are experimentally and numerically evaluated. These acoustic liners are associated with different open-area-ratios, thus enable the porosity effect to be studied. Controllable joint bias-grazing flow is applied to the liners. This simulates the flow configurations of a real engine system involving both bias and grazing flow. The damping performance of these liners is characterized by power absorption or transmission loss coefficient. Such coefficient is measured, as the tonal noise frequency is changed from 200 to 800 Hz. It is shown that the grazing flow can reduce the maximum power absorption coefficient, while the bias flow can improve the liners damping performance. The double-layer liner is associated with a larger maximum power absorption and a broader frequency range in comparison with that of the single-layer one. Finally, it is found that when the open area ratios of the inner and outer liners are approximately 1.1%, the damping performances of these liners are dramatically reduced, especially at higher frequency range. Increasing the open area ratios of the inner or outer liner leads to the maximum power absorption and the effective frequency range being dramatically improved.

Session 2pMU

Musical Acoustics: Pitch, Dynamics, and Vowel Tuning in Choral Voice

Ingo R. Titze, Chair
National Center for Voice and Speech, 136 South Main Street, Suite 320, Salt Lake City, UT 84101-3306

Chair's Introduction—1:00

Invited Papers

1:05

2pMU1. Maximizing the acoustic benefits of vocal vibration and resonance in choral singing. Laurier A. Fagnan (Campus Saint-Jean, Univ. of AB, 4915 - 114 B St., Edmonton, AB T6H 3N2, Canada, Lfagnan@ualberta.ca)

As an artistic discipline, choral singing often shies away from fully exploiting the vocal and acoustic benefits of vibration and resonance. It is sometimes thought that those two properties are more germane to the solo voice than to ensemble singing where "sameness" is often lauded above all else and *blend* can easily turn to *bland*. As Berton Coffin aptly states, "there is no reason to have a Stradivarius sound like a cigar-box violin so that both will sound the same." This session will show how the Stradivarius, complete-timbre-energy approach espoused by the old masters of the *bel canto* manner of singing is appropriate and beneficial to the art of choral singing. It will highlight the many acoustic advantages a choir can enjoy by applying the principles of regular, focused vibration and balanced, *chiaroscuro* (bright-dark) resonance throughout the voice's range and dynamic extent. Analyses such as high-speed videostroboscopy and real-time spectrography will help to better define and bridge the scientific and artistic aspects of energized choral sound. It will be shown that the proper application of *bel canto* principles has a positive effect on spectral balance, notably the presence of Singers' Formant energy, as well as on intonation and text intelligibility.

1:35

2pMU2. Pitch, dynamics, and vowel tuning in choral voice: Utilizing resonance strategies to train stylistic variance for choral singers. Jonathan D. Harris and Laurel Mehaffey (Music, College of the Holy Cross, 1 College St #195A, Worcester, MA 01610, jdharris@holycross.edu)

Building off of the notion of absolute timbre and its influence on the perceptual and acoustical definition of vowel, we explore acoustic resonance strategies that can help singers address the challenges of performing in varied styles. Choral singers have to manage variation in singing styles more than any other musician, and yet, they are often the musicians with the least amount of formal training. Most choral directors choose a one-size-fits-all resonance strategy that accentuates the fundamental and suppresses other harmonics. This "Choral Cathedral" resonance strategy minimizes the challenges inherent in balancing voices whose power spectra vary, but eliminates many of the possibilities available to the voice. By training singers to recognize resonant strategies, and how they function and feel, singers can begin to explore more of their voice's potential. Ultimately, this creates a more satisfying artistic experience for singer and audience alike. We use spectral analysis of different resonant strategies to show the impact of the absolute timbre of individual harmonics. We show how the vocal tract, even beyond the first and second formant, can create predictable acoustical environments unique to certain singing styles that singers can learn to achieve.

2:05

2pMU3. Loudness range of a choir based on choir size and voice range profiles of individuals. Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, 156 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@utah.edu)

The size of a choir is based on many factors, including availability of singers, size of the performance hall, economics of practice and performance, and the desired ensemble sound. It is generally assumed that a large choir has a greater dynamic range than a small choir. It is also generally assumed that a choir with well-trained singers can produce a greater dynamic range. A theoretical study is presented here that combines sound pressure level and loudness of homogeneous and non-homogeneous groups of singers to produce an overall sound level profile of a choir. Internal constraints are self-to-other ratio and individual voice range profiles of choir members. Results indicate that the dynamic range of a choir is determined mostly by the dynamic range of individuals with wide ranges, assuming that inhomogeneity is allowed. Choir size makes little difference. Singers with small dynamic ranges have little effect on choir dynamics unless louder voices are selectively turned off in pianissimo passages.

2:35

2pMU4. Pitch control in professional and amateur singers. Pasquale Bottalico (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Lansing, MI 48910, pb@msu.edu), Simone Graetzer, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

One aspect of major relevance to singing is the control of fundamental frequency. Ten amateur and ten professional singers sang two-octave arpeggi with three different levels of external auditory feedback, two tempi and two articulations (legato or staccato). The effects on pitch inaccuracy (defined as the distance in cents between the reference note and the sung note) of the following conditions were evaluated: (1) level of external feedback, (2) tempo (slow or fast), (3) articulation (legato or staccato), (4) tessitura (low, medium, or high), and (5) semi-phrase direction (ascending or descending). It was observed that inaccuracy was greatest in the descending semi-phrase arpeggi produced at a fast tempo and with a staccato articulation, especially for non-professional singers. The magnitude of inaccuracy was also relatively large in the high tessitura relative to the low and medium tessitura for such singers. Counter to predictions, when external auditory feedback was strongly attenuated or canceled by the hearing protectors, non-professional singers showed greater pitch accuracy than in the other external feedback conditions, which indicates the importance of internal auditory feedback in pitch control. With an increase in training, the singer's pitch inaccuracy decreases.

2:50

2pMU5. The effect of *chiaroscuro* and *coup de glotte* training on reducing the effects of intrinsic pitch in sung vowel transitions. Grace H. Crosby, Benjamin V. Tucker, and Laurier Fagnan (Linguist, Univ. of AB, 116 St. & 85 Ave., Edmonton, AB T6G 2R3, Canada, gcrosby@ualberta.ca)

Intrinsic pitch is an acoustic value that differs depending on vowel height. Sapir (1989) describes the various theories that provide an explanation for intrinsic pitch. Essentially, vowels which are produced higher in the filter (e.g., /i/ and /u/) tend to be pitched slightly higher than vowels produced at lower locations in the filter (e.g., /a/). The purpose of this study is to determine whether the *chiaroscuro* and *coup de glotte* vocal techniques used in the *bel canto* method of singing reduce the natural effects of intrinsic pitch of sung vowels by attempting to equalize both resonance and vibrational properties across the vowel spectrum. Six singers (2 sopranos, 2 tenors, and 2 basses) were recorded in this study. Participants sang a set of vowel transition exercises in a pre-training setting, then were trained in *chiaroscuro* and *coup de glotte* techniques, after which they did a post-training recording. Pitch, formant values, and amplitude (as mean values and at various times within each vowel) were extracted for a before versus after comparison. The results will be discussed in relation to how this technique might be helpful in reducing the effect of intrinsic pitch on sung vowels.

3:05

2pMU6. How singing experience affects the "brightness" of humming? Ae Na Yang, Marina Takabayashi, Sachi Itagaki, Hayato Kikuchi, and Kota Kobayasi (Doshisha Univ., Kyoto, Kyotanabe-City, Kyoto Tatara Miyako Tani 1-3, Japan, bmm1121@mail4.doshisha.ac.jp)

During singing practice, people are often told to sing in bright voice to improve singing performance. The purpose of this study is to determine how singing practice affects the acoustical characteristics and expressiveness of "brightness." In this study, eight singers (having experience in chorus for 0 to 13 years) sang three types of humming with G4 (392 Hz). Singers were told to sing either "bright," "dark," or "normal" humming. Subjects ($n = 16$) listened to different types of humming in pairs, and judged which was brighter. The results showed that the subjects perceived "bright" humming as bright. As singers practiced longer, their "brightness" was more expressive. Acoustical analysis revealed that fundamental frequencies of "bright" humming were higher than dark humming by 3 Hz, and that bright humming had higher amplitude in spectrum from 4900 to 6400 Hz by 4.5 to 6.5 dB. In the following experiment, subjects listened to and judged the brightness of humming, whose amplitude in the range was systematically amplified. As a result, the more amplified the humming had in the range, the more likely subjects evaluated it as bright. Our results suggest that singers can obtain the ability to express "brightness" by producing these acoustical characteristics through practicing singing.

Session 2pNS

Noise and Signal Processing in Acoustics: Statistical Learning Techniques in Noise Research

Jonathan Rathsam, Cochair
NASA Langley Research Center, MS 463, Hampton, VA 23681

Edward T. Nykaza, Cochair
ERDC, 2902 Newmark Drive, Champaign, IL 61822

Chair's Introduction—1:30

Invited Papers

1:35

2pNS1. Deep learning for unsupervised feature extraction in audio signals: A pedagogical approach to understanding how hidden layers recreate, separate, and classify audio signals. Edward T. Nykaza (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil), Arnold P. Boedihardjo (ERDC-GRL, Alexandria, VA), Zhiguang Wang, Tim Oates (Comput. Sci. and Elec. Eng., UMBC, Baltimore, MD), Anton Netchaev, Steven L. Bunkley (ERDC-ITL, Vicksburg, MS), and Matthew G. Blevins (ERDC-CERL, Champaign, IL)

Deep learning is becoming ubiquitous; it is the underlying and driving force behind many technologies we use everyday (e.g., search engines, fraud detection warning systems, and social-media facial recognition algorithms). Over the past few years, there has been a steady increase in the number of audio and acoustics related applications of deep learning. But what is exactly going on under the hood? In this paper, we focus on deep learning algorithms for unsupervised feature learning. We take a pedagogical approach to understanding how the hidden layers recreate, separate, and classify audio signals. We begin with a simple pure tone dataset, and systematically increase the complexity of this dataset in both frequency and time. We end the presentation with some feature extraction examples from real-world environmental recordings, and find that these features are easier to interpret given the understanding developed from the simpler tone datasets. The unsupervised feature learning techniques explored in this paper include: restricted Boltzmann machines (RBMs) and auto-encoders (AEs).

1:55

2pNS2. The importance of feature selection in supervised machine learning problems in acoustics. Eric A. Dieckman (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, edieckman@newhaven.edu)

As machine learning processes begin to gain traction and are applied to problems in acoustics, there is a need to better understand the importance of all stages of the process. Here, we discuss the preprocessing stage, where features are extracted from raw data and the best features are chosen using algorithms such as linear discriminant analysis (LDA). This allows the creation of a small-dimensional, information-dense feature vector to be used in the machine learning process. Applications discussed include classification of ground vehicles based on their acoustic backscatter signature and classification of wall structures based on transmission loss measurements.

2:15

2pNS3. Machine-learning models for the prediction of long-range outdoor sound propagation. Carl R. Hart, D. K. Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, carl.r.hart@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), and Edward T. Nykaza (U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Long-range outdoor sound propagation is characterized by a large variance in sound pressure levels due to factors such as refractive gradients, turbulence, and topographic variations. While conventional numerical methods for long-range propagation address these phenomena, they are costly in computational memory and time. In contrast, machine-learning algorithms provide very fast predictions, which this study considers. Observations from either experimental data, or surrogate data from a numerical method, are required for the training of machine-learning models. In this study, a comprehensive training set for the machine learning was created from excess attenuation predictions made with a Crank-Nicholson parabolic equation (CNPE) model. Latin hypercube sampling of the parameter space (source frequency, meteorological factors, boundary conditions, and propagation geometries) generates a set of input for the CNPE model and machine-learning models. Consideration is given to ensemble decision trees, ensemble neural networks, and cluster-weighted models for nonlinear regression. The large variance in excess attenuation, from the CNPE model, presents a challenge for accurate machine-learning model predictions. For example, given 5000 samples the overall root-mean-square error for an ensemble decision tree model is 6.7 dB. Errors related to sample size, modeling approaches, and propagation ranges are quantified in this study.

2:35

2pNS4. On the distribution of impulsive sound events for environmental noise assessment. Frank Van den Berg and Frits Van der Eerden (TNO, Oude Waalsdorperweg 63, The Hague 2597 AK, Netherlands, frank.vandenberg@tno.nl)

The noise levels of impulsive sounds are subject to variation, mostly due to changes in the meteorological situation which have a strong influence on the noise propagation. For environmental noise assessment studies the variation in the single events levels as well as the long term (averaged) level should be considered. To calculate these one can use a number of variations for the atmospheric absorption and the excess attenuation, alongside with histograms of for instance the wind speed and wind directions. Just recently, a new approach has been enforced in the Netherlands to assess the sound from shooting ranges. Details on this calculation method will be given in which meteorological classes are used to account for varying wind speed and temperature gradients. This method makes it also possible to describe the distribution of occurring sound exposure levels around shooting ranges. The distribution of impulsive sounds is further illustrated with data obtained from projects carried out on: long range propagation of high-energy blasts, monitoring military training areas with detonations and muzzle blast noise, and monitoring fireworks in an urban area.

2:55–3:10 Break

3:10

2pNS5. Trans-dimensional Bayesian approaches to room acoustic modal analysis. Douglas Beaton and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, beatod@rpi.edu)

In noise control engineering, it is often desirable to assess the modal behavior of mechanical structures or a room and determine what implications any modes present in the system under investigation will have on noise. The application of Bayesian methods to analyze modal behavior in experimentally measured room impulse responses is of main interest in the current work. These methods extract relevant information from the room impulse response in the time domain and can be used to estimate modal parameters such as mode frequency, amplitude, decay time, and phase. Previous efforts in Bayesian modal analysis divided the analysis into two distinct steps: model selection (determining the appropriate number of modes), and parameter estimation (determining the parameters of each mode). This work considers approaches that combine the two steps into a single trans-dimensional operation. The number of modes in a given model is defined as a parameter of that model. By doing so, the task of model selection becomes part of the parameter estimation, which itself becomes a trans-dimensional problem. Approaches to solve this trans-dimensional problem are applied to both simulated and experimentally measured room impulse responses. Results are compared with other Bayesian approaches and also with conventional Fourier analysis.

3:30

2pNS6. Statistical analysis of multilayer porous absorbers with Bayesian inference. Cameron J. Fackler (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, facklc@alum.rpi.edu), Alistair Hurrell (Dept. of Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom), Douglas Beaton, and Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

In noise control and other applications, porous sound-absorbing materials may be constructed of multiple layers of homogeneous materials. This work applies Bayesian inference to study and analyze such multilayer porous materials. The analysis utilizes a measurement of the overall material's acoustic surface impedance and a transfer-matrix porous material acoustic model. The number of layers present in the multilayer porous absorber under test and the physical properties of each layer are inversely determined. Bayesian model selection implements Occam's razor to determine the number of layers present in the sample, based on the measured impedance data and without any a priori knowledge of the number of layers. Once the number of layers has been determined, Bayesian parameter estimation inversely determines the physical properties of all layers simultaneously. A Markov chain Monte Carlo method, nested sampling, is applied to explore efficiently the high-dimensional parameter space inherent to this inverse problem. This sampling method automatically implements the model selection and parameter estimation components to analyze practical multilayer porous absorbers.

3:50

2pNS7. Dose-effect relationships for annoyance due to road traffic noise: Multi-level regression and consideration of noise sensitivity. Laure-Anne Gille (Direction territoriale Ile de France, Cerema, 21-23 rue Miollis, Paris Cedex 15 75732, France, laure-anne.gille@cerema.fr) and Catherine Marquis-Favre (Univ Lyon, ENTPE, Laboratoire Génie Civil et Bâtiment, Vaulx-en-Velin, France)

An *in situ* survey was performed in eight French cities in 2012 to study the annoyance due to combined transportation noise sources. The European Union dose-effect relationships were compared to these new survey data for noise annoyance due to road traffic noise. The measured annoyance was not satisfactorily predicted by these curves: only the percentages of people highly annoyed by road traffic noise was well predicted. Following a multi-level regression as used to construct the European Union dose-effect relationships, new dose-effect relationships were proposed. These new dose-effect relationships enabled a better calculation of noise annoyance due to road traffic noise. Finally, a methodology to consider noise sensitivity in the computation and the percentage of people sensitive to noise in the results is proposed, as this non-acoustical factor is well known to influence noise annoyance. However, the results showed that taking into account such variable did not enable to enhance the dose-effect relationships.

4:10

2pNS8. Uncertainty estimates of psychoacoustic thresholds obtained from group test. Jonathan Rathsam and Andrew Christian (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jonathan.rathsam@nasa.gov)

A common goal in laboratory psychoacoustic testing is determining the level, or threshold, at which a test signal is judged subjectively equivalent to a reference signal. Adaptive methods, in which the next signal level depends on the response to the previous signal, are most efficient and precise for determining thresholds but only accommodate one subject at a time. At NASA, testing one subject at a time to achieve a sample size representative of a larger population would exhaust testing resources. Instead, three or four subjects are tested simultaneously using preselected signal levels. When a threshold is estimated from a curve fit through group test data, techniques are required for assessing the threshold uncertainty. In this presentation, we examine the Delta Method, the Generalized Linear Model (GLM), the Nonparametric Bootstrap, and Bayesian Posterior Estimation (BPE). Each technique is first exercised on a manufactured, theoretical dataset. When we are confident we are using the methods correctly, we apply them to two psychoacoustic datasets. The Delta Method is the simplest to implement. The BPE is the most versatile, allowing the inclusion of prior information. While useful, the Nonparametric Bootstrap takes longer to calculate. The GLM is found to be least robust.

4:30–4:50 Panel Discussion

TUESDAY AFTERNOON, 24 MAY 2016 — SALON H, 1:00 P.M. TO 4:10 P.M.

Session 2pPA

Physical Acoustics and Biomedical Acoustics: Vortex Beams and Radiation Torque Physics II

Philip L. Marston, Cochair
Physics and Astronomy Dept., Washington State University, Pullman, WA 99164-2814

Likun Zhang, Cochair
University of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199

Invited Papers

1:00

2pPA1. Numerical calculation of acoustic radiation force and torque on non-spherical particles in Bessel beams. Felix B. Wijaya and Kian Meng Lim (Mech. Eng., National Univ. of Singapore, Dynam. Lab E1 02-03 National University of Singapore 1 Eng. Dr. 2, Singapore 117576, Singapore, a0107285@u.nus.edu)

Manipulation of biological cells using acoustic radiation force has drawn a lot of attention in recent years. The force and torque acting on cells are usually estimated from analytical and semi-analytical solutions derived for simple shapes, such as spheres and ellipsoids, typically in an axisymmetric configuration. Since biological cells come in various shapes and sizes and they may have an arbitrary orientation in a microfluidic channel, there is a need for a more versatile and robust numerical model for evaluating the acoustic radiation force and torque. Motivated by this, a three-dimensional boundary element model is developed for calculating radiation force and torque on particles of arbitrary shapes and sizes subjected to arbitrary acoustic waves. The first order acoustic field is solved by using the boundary element method. The second-order, time-averaged tractions are then obtained from the first order field. Subsequently, the resultant radiation force and torque are calculated by integrating the tractions over the surface of a fictitious sphere that encapsulates the particle. The force and torque on non-spherical particles subjected to acoustic Bessel beams are obtained using this numerical model. The effects of the beam cone angle and particle orientation on the radiation force and torque are investigated.

1:25

2pPA2. Acoustic forces acting on particles and fluids in microscale acoustofluidics. Henrik Bruus and Jonas T. Karlsen (Dept. of Phys., Tech. Univ. of Denmark, DTU Phys., Bldg. 309, Kongens Lyngby 2800, Denmark, bruus@fysik.dtu.dk)

Acoustofluidics relying on acoustic forces to handle fluids and particles in microfluidic systems has emerged as a useful tool for characterizing, focusing, separating, and sorting cells based on their acousto-mechanical properties. Here, we present recent advances in the theoretical understanding of acoustic forces on particles and fluids. In particular, we address the effects of thermoviscous boundary layers on acoustic scattering off sub-micron particles or droplets. Re-examining the far-field method of calculating acoustic radiation forces and torques, we show that exact non-perturbative expressions can be derived regardless of boundary layer dissipation. The necessary condition for moving the surface of integration from the particle surface to the far field, is the time-periodicity of the system rather than negligible boundary-layer dissipation. In the long-wavelength limit, this approach leads to particularly simple expressions for the force and torque acting on a particle in a thermoviscous fluid. Finally, relaxing the requirement of having two immiscible phases (particle/fluid or droplet/fluid), we generalize the theory to include acoustic forces acting on continuous density and compressibility distributions of inhomogeneous fluids, such as aqueous salt solutions.

1:50

2pPA3. Singular acoustics: Transfer of orbital angular momentum to matter. Régis Wunenburger (Univ. of Paris 6, Paris, France) and Etienne Brasselet (Univ. of Bordeaux, LOMA (UMR5798), 351 cours de la libération, Talence 33400, France, etienne.brasselet@u-bordeaux.fr)

A general feature of wave physics is the existence of phase singularities. One usually refers to beams carrying phase singularities as vortex beams, which have become more popular in the field of optics than in acoustics. In particular, the first experimental demonstration of transfer of orbital angular momentum of sound to matter came much later than its optical counterpart. Here, we will review our recent results regarding the transfer of acoustic orbital angular momentum to matter in the ultrasonic domain. This includes (i) the quantitative test of acoustic orbital angular momentum transfer to a sound absorbing object, (ii) the introduction of a novel phenomenon named "rotational acoustic streaming," and (iii) the demonstration a nondissipative sound-matter orbital angular momentum transfer mediated by chiral scattering.

2:15

2pPA4. On-chip generation of acoustical vortices with swirling surface acoustic waves for single particle manipulation and vorticity control. Antoine Riaud, Michael Baudoin (IEMN, Université de Lille 1, IEMN, Ave. Poincaré, Villeneuve d'Ascq 59652, France, michael.baudoin@univ-lille1.fr), Jean-Louis Thomas (INSP, Sorbonne universités, Univ. Paris 6, Paris, France), and Olivier Bou Matar (IEMN, EC Lille, Villeneuve d'Ascq, France)

Surface acoustic waves (SAWs) are versatile tools for the manipulation of fluids at small scales. These waves can be used to displace, divide, merge, and atomize sessile droplets, but also actuate fluids embedded in microchannels. In this presentation, we will show that IDTs array and inverse filter technique enable on-chip synthesis of a new type of SAWs, called swirling SAWs, which degenerate into bulk acoustical vortices when transmitted to a liquid. These acoustical vortices can be tailored to create a 3D particle trap and thus selective single particle acoustical tweezers, with digital control of the trap position. They can also induce controlled vortical flows, whose topology essentially relies on the topology of the underlying acoustical vortex.

2:40–2:55 Break

2:55

2pPA5. Acoustic radiation and viscous torque for micromanipulation controlled rotation of particles in fluid cavities. Andreas Lamprecht, Thomas Schwarz, Jingtao Wang, and Jurg Dual (Dept. of Mech. and Process Eng., ETH Zurich, Tannenstrasse 3, CLA H23.1, Zurich, Zurich 8092, Switzerland, lamprecht@imes.mavt.ethz.ch)

Our investigations at ETH Zurich aimed at the theoretical analysis of the acoustic torque and its experimental realization of a controlled rotation of spherical and non-spherical particles by ultrasound. Ultrasonic manipulation of particles provides a contactless handling method for particles suspended in a fluid by acoustic streaming and radiation forces. In addition to the translation of particles in all three spatial directions, particles like functional beads, cells, clumps of cells, fibers, etc., can be rotated. Various methods for the rotation of non-spherical particles were developed with the acoustic radiation torque. The necessary varying pressure field, where the orientation of the nodal pressure lines was controlled by two orthogonal standing waves, was achieved by a modulation of one single parameter over time, e.g., amplitude, phase, and frequency. Stable particle and fiber rotations up to 40 rpm were reached. The rotation can be performed continuously or in a stepwise fashion. Moreover, the rotation of spherical objects was realized by the viscous torque. This torque is formed by acoustic streaming, due to two orthogonal standing waves shifted in phase at the same excitation frequency and amplitude. Angular rotations up to 1200 rpm for spherical 35.5 μm copolymer particles were reached.

Contributed Papers

3:20

2pPA6. Acoustic propagation from a helicoidal wavefront source in an ocean environment. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

A helicoidal wavefront source generates a pressure field with a phase dislocation along its axis. About this dislocation, the phase varies linearly with angle while at the dislocation there is a null in the pressure field. The simplest source which can produce this field consists of two crossed dipoles driven 90 degrees out of phase with one another. This phased dipole source was previously used to model propagation from a spiral wavefront beacon in an ocean environment (Hefner and Dzikowicz, JASA, 2012). For the spiral wavefront beacon, the dislocation axis is aligned vertically and the phase variation near the x-y plane is used for underwater navigation. This phased dipole source was shown to be related to the point source through a simple transform which made it possible to transform any point source solution in an ocean environment into the solution for a spiral source in the same environment. This transformation is generalized such that the dislocation axis can be oriented in any direction. This makes it possible to examine helicoidal wavefront propagation in an ocean environment as well. Applications of this general transformation are presented for a helicoidal source near the sea surface and in an ocean waveguide.

3:35

2pPA7. Radiation force expansions in terms of partial wave phase shifts for scattering: Examples. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Likun Zhang (Phys. Dept. & Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, Austin, TX)

When evaluating radiation forces on spheres in acoustic beams, with or without helicity, the interpretation of analytical results is greatly simplified by retaining the use of s-function notation for partial-wave coefficients imported into acoustics from quantum scattering theory in the 1970s. This facilitates easy interpretation of various efficiency factors [L. Zhang and P. L. Marston, Phys. Rev. E. **84**, 035601 (2011); L. Zhang and P. L. Marston, Bio. Opt. Express **4**, 1610–1617 (2013); (E) **4**, 2988 (2013)]. For situations in which dissipation is negligible each partial wave s-function becomes characterized by a single parameter: the partial wave phase shift that parameterizes possible degrees of freedom [P. Marston and L. Zhang, J. Acoust. Soc. Am. **131**, 3534 (2012); P. L. Marston, J. Quant. Spectrosc. Radiat. Transf. **162**, 8–17 (2015)]. The s-function and associated phase shifts are associated with scattering by plane traveling waves, and the incident wavefield of interest is separately parameterized. (When considering allowed outcomes, the method of fabricating symmetric objects having a desirable set of plane-wave scattering partial-wave phase shifts becomes a separate issue.) The present analysis illustrates the advantages of the formulation by extending some prior expansions associated with radiation forces. [Supported in part by ONR.]

3:50–4:10 Panel Discussion

TUESDAY AFTERNOON, 24 MAY 2016 SALON D, 1:00 P.M. TO 3:55 P.M.

2p TUE. PM

Session 2pPPa

Psychological and Physiological Acoustics: Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical and Physiological Collaborations

Anna C. Diedesch, Cochair
Hearing & Speech Sciences, Vanderbilt University, 7012 Sonya Drive, Nashville, TN 37209

Adrian KC Lee, Cochair
University of Washington, Box 357988, University of Washington, Seattle, WA 98195

Chair's Introduction—1:00

Invited Papers

1:05

2pPPa1. Differences in cortical responses to spatial changes in foreground versus background auditory objects. Darrin K. Reed, Brigitta Tóth, and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, dkreed@bu.edu)

Identifying neural markers related to the integration of auditory features—such as spectrum, temporal coherence, and lateral position—can help to reveal the mechanisms underlying how the auditory system processes simultaneous sound sources. For a stimulus that is composed of randomly chosen successive inharmonic tone complexes, a subset of tones can perceptually segregate from the other simultaneous tones if the subset repeats for a sufficient number of tokens. Listeners can reliably detect this repeating pattern as a figure sound object amidst a randomly changing background. An electroencephalogram was recorded while listeners performed the detection of changes in interaural time difference (ITD) cues that were attributed to either the figure or the background. Detection of ITD changes for either auditory object elicited a fronto-central early negativity (100–200 ms) and P300 event-related potentials (ERP) associated with identification of a behaviorally relevant event. However, the ERP amplitudes were larger for the ITD changes associated with the figure compared those associated with an ITD change of the background. This illustrates that the manner in which spectro-temporal features combine to form auditory objects can modulate cortical responses to spatial information. The result supports the notion that spatial features integrate subsequent to object formation.

1:30

2pPPa2. Shining a light on the neural signature of effortful listening. Ian M. Wiggins, Pramudi Wijayasiri, and Douglas Hartley (National Inst. for Health Res. Nottingham Hearing Biomedical Res. Unit, Div. of Clinical Neurosci., School of Medicine, Univ. of Nottingham, Ropewalk House, Nottingham NG1 5DU, United Kingdom, ian.wiggins@nottingham.ac.uk)

This study used functional near-infrared spectroscopy (fNIRS) to investigate a possible neural correlate of effortful listening. Hearing-impaired individuals report more listening effort than their normally hearing peers, with negative consequences in daily life (e.g., increased absenteeism from work). Listening effort may reflect neuro-cognitive processing underlying the recovery of meaning from a degraded auditory signal. Indeed, evidence from functional magnetic resonance imaging (fMRI) suggests that the processing of degraded speech is associated with increased activation in cortical areas outside the main auditory regions within the temporal lobe. However, acoustic scanner noise presents a serious methodological challenge in fMRI. This challenge can potentially be overcome using fNIRS, a silent, non-invasive brain-imaging technique based on optical measurements. In the current study, we used fNIRS to confirm the finding that attentive listening to degraded (noise-vocoded) speech leads to increased activation in the left inferior frontal gyrus (LIFG), compared to a clear-speech control condition. In contrast, robust activation in the auditory cortices was unaffected by speech clarity or attentional focus. Our results support the suggestion that the LIFG plays a role in compensating for degradation to the speech signal. Furthermore, fNIRS may hold promise as a flexible tool to examine the neural processes underlying effortful listening.

1:55

2pPPa3. Temporary unilateral hearing loss during development impairs behavioral and neural sensitivity to interaural level difference cues for sound localization. Kelsey L. Anbuhl (Neurosci. Training Program, Univ. of Colorado Anschutz Medical Campus., RC1-N, Rm. 7401G, 12800 East 19th Ave., Aurora, CO 80045, kelsey.anbuhl@ucdenver.edu), Nathaniel T. Greene, Andrew D. Brown, Victor Benichoux (Physiol. and Biophys., Univ. of Colorado Anschutz Medical Campus., Aurora, CO), Alexander T. Ferber (Neurosci. Training Program, Univ. of Colorado Anschutz Medical Campus., Aurora, CO), and Daniel J. Tollin (Physiol. and Biophys., Univ. of Colorado Anschutz Medical Campus., Aurora, CO)

Children who experience persistent conductive hearing loss (CHL) early in life often display binaural hearing impairments that persist long after CHL is resolved, suggesting abnormal central auditory development. Abnormal sensitivity to interaural level differences (ILDs) is particularly likely as a CHL (such as an ear infection) can attenuate sound in the affected ear by $>$30 dB, dramatically distorting ILD cues. Here, we quantified the effects of unilateral CHL on (1) behavioral spatial acuity and (2) neural information processing of ILD cues in the guinea pig auditory midbrain (inferior colliculus, IC) using the mathematical framework of Fisher information (FI). Animals raised with unilateral CHL displayed larger minimum audible angles for high-pass noise compared to age-matched controls, suggesting impaired ILD sensitivity. Based on acoustic directional transfer function measurements, ILD discrimination thresholds were elevated by ~3–6 dB. Following behavior, extracellular recordings were made in the IC contralateral to the previously occluded ear, and ILD discrimination thresholds for single neurons were determined using FI. Across the population, neural ILD discrimination was moderately impaired (~2–3 dB worse-than-control) in CHL animals. Impaired processing of ILD in the IC may in part explain the spatial discrimination deficits observed in animals and children with developmental CHL.

2:20–2:40 Break

2:40

2pPPa4. Removing effects of ear-canal acoustics from measurements of otoacoustic emissions. Karolina Charaziak and Christopher Shera (MEE\HMS, 243 Charles St., Boston, MA 02114, KarolinaCharaziak2013@u.northwestern.edu)

Otoacoustic emissions (OAEs) are the acoustic fingerprints of the inner ear—when carefully measured in healthy ears, their spectra, although highly individualized, remain stable over time. Thus, OAE changes usually indicate changes in cochlear function, e.g., due to efferent modulation, aging, noise trauma, and/or exposure to harmful agents. In humans, however, the reproducibility of OAE measurements is compromised by ear-canal standing waves at relevant frequencies. We show that even when stimulus levels are tightly controlled using methods designed to avoid standing-wave problems (forward-pressure-level calibration), distortion-product (DP)OAE levels vary by ~10–15 dB near half-wave resonant frequencies, depending on probe insertion depth (deep versus shallow). We propose a method, derived from a tube model of the ear canal, that separates the initial outgoing OAE pressure wave at the eardrum from reflected OAEs trapped in the residual ear-canal space. The emitted pressure level (EPL) represents the load-independent OAE level that would be recorded in an ideal, reflectionless canal. When DPOAE levels are converted to EPL, variability across insertion depths decreases by ~10 dB near half-wave resonant frequencies. EPL may provide a simple way to reduce confounding OAE variability across subjects and to improve the reliability of OAE measurements for detecting cochlear changes.

3:05

2pPPa5. Modeling dynamic properties of spontaneous otoacoustic emissions: Low-frequency biasing and entrainment. Dario Vignali, Stephen J. Elliott, and Ben Lineton (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, dv206@soton.ac.uk)

Spontaneous otoacoustic emissions (SOAE) are sounds generated inside the living cochlea and are regarded as by-products of the active mechanism present in the peripheral auditory system. There is still debate whether these emissions are the product of local oscillators located at various positions along the cochlea (local oscillator model) or if they result from standing waves due to a global collective phenomenon that involves different aspects of the cochlea (global standing wave model). This paper uses a global standing wave cochlear model to predict various features of SOAEs. This involves a state-space formulation with a spatially distributed set of nonlinear active micromechanical elements coupled via cochlear fluid coupling. Simulation results have been compared with available experimental data and demonstrate two interesting nonlinear features of the cochlea: first, nonlinear properties of SOAEs modulated by external low-frequency bias tones are easily predicted and can be used to investigate the plausibility of different nonlinear functions incorporated in the micromechanical elements. Second, entrainment patterns can be obtained when the model is stimulated by a swept-tone: results

show distinct areas of beating and others of entrainment between the external stimulus and the SOAE, which depend on the level and instantaneous frequency of the sweep.

3:30

2pPPa6. Hidden hearing loss in tinnitus with normal hearing thresholds. Brandon T. Paul (Psych., Neurosci., & Behaviour, McMaster Univ., 1280 Main St. West, PC 333, Hamilton, ON L8S 4K1, Canada, paulbt@mcmaster.ca), Ian Bruce (Elec. and Comput. Eng., McMaster Univ., Hamilton, ON, Canada), and Larry Roberts (Psych., Neurosci., & Behaviour, McMaster Univ., Hamilton, ON, Canada)

Tinnitus—the phantom ringing of the ears—is thought arise from neuroplastic changes in the central auditory system in response to peripheral hearing damage. However, a minority of tinnitus sufferers have clinically normal hearing thresholds. One explanation of these cases is that auditory nerve fibers with low firing thresholds (LT-ANFs) are intact, but ANFs with high thresholds (HT-ANFs) are not. HT-ANF damage would be "hidden" to the audiogram but evident in suprathreshold tests. To test this hypothesis, we measured the ability of tinnitus and control subjects with normal audiograms to detect amplitude modulation (AM) in a 5 kHz, suprathreshold tone in a narrowband noise. We also recorded by 32-channel EEG the "envelope following response" (EFR, generated subcortically) to the same AM tone in conditions of noise and no noise. Tinnitus subjects had worse AM detection thresholds and had smaller EFRs compared to controls. Simulations of ANF responses from the model of Zilany *et al.* (2014) found that in addition to ~100% loss of HT ANFs, a further ~30% loss of LT fibers was needed to account for the reduced EFRs of tinnitus subjects. ~30% of LT ANFs would not have been expected to affect hearing thresholds. [Work supported by NSERC of Canada.]

TUESDAY AFTERNOON, 24 MAY 2016 SALON H, 4:25 P.M. TO 5:30 P.M.

Session 2pPPb

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Robert P. Carlyon, Chair
CBU, MRC, MRC CBU, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom

Chair's Introduction—4:25

Invited Paper

4:30

2pPPb1. Bridging the chasm: Animal physiology and human psychophysics. Alan Palmer (MRC Inst. of Hearing Res., Univ. Park, Nottingham NG7 2RD, United Kingdom, alan@ihr.mrc.ac.uk)

A goal of animal auditory neuroscience is to understand how humans process and perceive sound. Encouragingly, hypotheses about brain function, generated from psychoacoustics, are sometimes confirmed by physiological experiments. A notable example is the Jeffress model of processing of interaural time differences. This has been largely supported by physiology and indeed we showed that neural activity in the midbrain and cortex, in response to Binaural Masking Level Difference stimuli, was entirely consistent with this cross-correlation model of binaural processing. However, such comparisons of physiology and psychophysics are often beset by caveats of species and anesthesia. This is one reason why the sharpness of human frequency tuning is still currently debated. In recent experiments, we have measured tuning behaviorally and by direct and indirect physiological methods, all in the same species. These different measures are in good agreement, supporting the notion that the cochlea determines perceptual frequency selectivity, and indirectly that perhaps frequency selectivity is different in humans. That notwithstanding, it seems clear that comparisons of animal physiology and human psychophysical measures provide important confirmations and insights into human auditory function.

Session 2pSAa

Structural Acoustics and Vibration: Real-World Instructive Case Studies in Structural Acoustics and Vibration

Robert M. Koch, Cochair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Elizabeth A. Magliula, Cochair
Division Newport, Naval Undersea Warfare Center, 1176 Howell Street, Bldg. 1302, Newport, RI 02841

Chair's Introduction—1:00

Invited Papers

1:05

2pSAa1. What would MacGyver do? Unconventional vibration testing for Mercy Sioux City helistop. Jon W. Mooney (Acoust. & Vib., KJWW Eng. Consultants, 623 26th Ave., Rock Island, IL 61201, mooneyjw@kjww.com)

A trip to Harbor Freight, a hardware store and some MacGyvering—these were essential to arriving at a vibration analysis for a hospital wanting to add a helistop to its roof. Normally not a challenging assignment, the analysis for this project was complicated by many factors, including: the existence of several functioning cath labs on the floor below the roof; unknown vibration and EMI sensitivity of the cath lab equipment; a limited budget— and only one week to prepare for the study. This presentation will explain in detail the goals and constraints of the project, and how three engineers—one an acoustician—developed relatively-simple-yet-ingenious, and low-cost, solutions.

1:25

2pSAa2. Practical aspects of design, tuning, and application of dynamic vibration absorbers. Vyacheslav M. Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vyacheslav.ryaboy@newport.com)

Theory of dynamic vibration absorbers is a classic topic that has been thoroughly researched. Nevertheless, implementing the dynamic system in a structure and applying it properly still meet some challenges that have not been adequately addressed in the technical literature. Some of practical problems include achieving thermal stability, building a damped device with the desired undamped frequency, providing mechanisms for tuning or adjusting the dynamic absorber *in situ*, and effective tuning of multiple absorbers. The paper describes a new patented design of dynamic vibration absorber for optical tables together with the new method of tuning. The tuning procedure uses only one sensor and does not require access to the moving mass. Applications include suppressing flexural resonance vibrations of standard optical tables and the recent project on vibration control of large support bench for unique petawatt laser system at Lawrence Berkeley National Laboratory.

1:45

2pSAa3. Far field prediction of sound pressure level using panel contribution analysis and scale modeling. David W. Herrin and Gong Cheng (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, dherrin@engr.uky.edu)

Panel contribution analysis is reviewed and then utilized to predict the sound radiation from a generator set. The generator set is divided into patches and transfer functions between the sound pressure at a point in the far field and the velocity of a patch were determined reciprocally both for the full-scale structure and a half-scale model. With the generator set running, a P-U probe was used to measure particle velocity and sound intensity simultaneously on each patch. The sound pressure level at a receiver point located 4.9 m away from the side of the generator set was calculated assuming uncorrelated and correlated sources. The predicted sound pressure, based on both full and half scale models, compared well with that measured. Transfer functions were also determined using boundary element methods and compared well with measurement. In addition, the contributions from each surface of the generator set enclosure were determined.

2:05

2pSAa4. The problem of the noisy golf club. Peter A. Kerrian and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

Player perception of the quality of a hand-held sports implement is strongly influenced by the sound generated at impact with a ball. A recent innovative golf club driver design increased the moment-of-inertia of the club, making it more forgiving to hits off center and allowing players to hit balls with greater accuracy. However, this club has not been readily accepted because players hate the loud and annoying sound it makes when striking a golf ball. In this paper, we will discuss the acoustic and vibrational properties of this club as measured through experimental modal analysis, laser Doppler vibrometry, and acoustic field tests. Vibration and acoustic data correlate well and reveal that the annoying high amplitude components in the impact sound are due to vibrational modes in the sole of the club head, at frequencies below that of the "trampoline mode" of the club face. Implications resulting from this study will influence future golf club designs to make them sound more pleasing without altering performance.

Contributed Papers

2:25

2pSAa5. Sound characteristic analysis of sound-pipe-breakwater on the east-coast in Gangneung South Korea. Hyungwoo Park, Won-Hee Lee, and Myung-Sook Kim (SoongSil Univ., 1212 Hyungham Eng. Bldg. 369 Snagdo-Ro, Dongjak-Gu, Seoul, Seoul 156743, South Korea, pphw@ssu.ac.kr)

There is mysterious sounds which can hear after 2010 the new breakwater construction, in a small port city "Yung-Jin" of Gangneung East Sea of Korea. In particular, the mysterious sounds are spread loud honk, when north-east wind are blown in fall and winter of the seasons. People, who have heard the sound that is regarded as a sinister sound, called that like "Kelpie sound," "Sea whistle," and "Pipe sound." This is an ordinary concrete breakwater structure, which has prevented the waves by rocks and tetrapods. The waves are usually considered wind-driven waves. The wind and waves were blocked in rocks and tetrapods. In this time, the wind causes a vortex by the tetrapods. The waterway of the inner harbor and the ocean filter a particular sound out. We can hear and analyze the Mysterious Sound of sound-pipe-breakwater, although the design of the breakwater is not to produce sound. In this study, we investigate and analyze the truth of the strange noises of breakwaters.

2:40

2pSAa6. Ultrasonic imaging with wave mode adaptive weights and global matched coefficient. Simone Sternini, Thompson V. Nguyen, and Francesco Lanza di Scalea (Structural Eng., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ssternin@eng.ucsd.edu)

This paper discusses some improvements to ultrasonic synthetic imaging in solids. Specifically, the study proposes new adaptive weights applied to the beamforming array that are based on the physics of the propagating waves, specifically the displacement structure of the propagating longitudinal (L) mode and shear (S) mode that are naturally coexisting in a solid. The wave mode structures can be combined with the wave geometrical spreading to better filter the array and improve its focusing ability compared to static array weights. The paper also proposes compounding, or summing, images obtained from the different wave modes to further improve the array gain without increasing its physical aperture. The wave mode compounding can be performed either incoherently or coherently, in analogy with compounding multiple frequencies or multiple excitations. Furthermore, the introduction of a global matched coefficient computed through the matching of measured and expected times of flight will be presented to show additional improvements in the image reconstruction process. Numerical simulations and experimental testing demonstrate the potential improvements obtainable by the wave structure adaptive weights and the global matched coefficient compared to either static weights in conventional delay-and-sum focusing, or adaptive weights based on geometrical spreading alone in minimum-variance distortionless response focusing.

2:55

2pSAa7. Preliminary study of extraterrestrial optical vibrometry for an impactor mission to Europa. Thomas Campbell, John Brewer, Masataka Okutsu, Diego Turo, and Joseph Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE., Washington, DC 20064, 96campbellt@cua.edu)

We present a methodology to determine the thickness and material properties of the icy crust of Europa, a satellite of Jupiter. Instruments proposed for NASA's Europa Multiple Flyby mission may be insufficient to adequately measure the ice's thickness. We propose a complementary experiment in which vibration of the Europan surface is induced by an artificial impact. Optical vibrometry techniques, such as 2-D digital image correlation and vibration magnification are considered for detecting the displacement field at a sub-pixel level. We make use of COMSOL Multiphysics, which simulates wave motions via the finite element method. Dispersion curves of elastic waves and the wavenumber filtering are used to estimate the thickness and properties of the extraterrestrial ice.

3:10

2pSAa8. A study on a sound fire extinguisher using special sound lens. Iksoo Ahn, Hyungwoo Park, Seonggeon Bae, and Myungjin Bae (TeleCommun. & Information, Soongsil Univ., 369 Sangdo-ro, Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com)

Sound fire extinguisher is developed based on principle of quenching fire by blocking inflow of oxygen with contacting vibration energy from low frequency sound under 100 Hz to fire and then, lowering its temperature. The first sound fire extinguisher made by a Federal Defence researcher was very huge and based on theory of initial principles, making it hard to be commercialized. Although next sound fire extinguisher developed by students of George Mason University was smaller than before, it was also at the basic stage with early theory. What is more, disturbance from the too long cable and inconvenience of having to carry all the tools were not effective to be commercialized. By improving these drawbacks, Sori Sound Engineering Research Institute (SSERI) of Soongsil University has developed new concept sound fire distinguisher. It concentrates sound through special sound lens and increases energy 10 times more to put out the fire. This study researches about reducing weight of sound fire extinguisher based on new concept special sound lens created by Sori Sound Engineering Research Institute of Soongsil University.

Session 2pSAb

Structural Acoustics and Vibration and Signal Processing in Acoustics: Wavenumber Transform Methods

Micah R. Shepherd, Chair
Applied Research Lab, Penn State University, PO Box 30, Mailstop 3220B, State College, PA 16801

Chair's Introduction—3:40

Invited Papers

3:45

2pSAb1. Wavenumber analysis of vibrating dimpled beams. Kyle R. Myers (Appl. Res. Lab., Penn State Univ., 3220B G Thomas Water Tunl, PO Box 30, State College, PA 16804, krm25@arl.psu.edu) and Koorosh Naghshineh (Mech. Eng. Dept., Western Michigan Univ., Kalamazoo, MI)

A structure's noise and vibration characteristics can be changed by introducing dimples onto the surface. These structural modifications are an attractive form of noise control since they are simple to manufacture and do not add mass to the structure. In order to gain a better understanding of the effect of dimples, beams with different number of dimples are considered in this study. The natural frequencies and mode shapes of beams with any number of dimples are computed using a boundary value model derived from Hamilton's Principle. The results of this model are compared to the finite element method and to previous literature. Then, the effect of the dimples on the radiation properties of beams is investigated by examination of their wavenumber spectra. By shifting the wavenumber content of the beam between supersonic and subsonic regions, dimples are able to change the amount of radiation emitted. For some boundary conditions, the wavenumber spectrum is time-dependent as the beam vibrates through a complete cycle. Several examples are presented that demonstrate how different numbers of dimples, as well as their locations and geometries, affect the wavenumber spectra of beams.

4:05

2pSAb2. Achievable performance of a novel Bragg-scattered acoustic receiving array. Benjamin Cray and Ivars Kirsteins (NUWC, 1176 Howell St., Newport, RI 02841-1708, benjamin.cray@navy.mil)

The merits of a novel beamforming technique, based on generating Bragg-scattered acoustic wavelengths along the surface of an array, will be described. The technique, denoted acoustic trace wavelength enhancement, relies on embedding periodic structures within an array, chosen to precisely replicate and shift an incident acoustic wavenumber into higher wavenumber regions. Thus, shorter trace wavelengths are created over the aperture surface. The enhancement technique is documented in two recent publications: enhanced directivity with array grating [J. Acoust. Soc. Am. **136**, 2014] and experimental verification of acoustic trace wavelength enhancement [J. Acoust. Soc. Am. **138**, 2015). These references dealt, however, solely with high signal-to-noise ratios. Specifically, we will investigate the noise characteristics of this new array by calculating its array gain and Cramer-Rao lower bounds on bearing estimation error for plane wave signals embedded in an isotropic Gaussian noise field. Of particular interest is how the performance of the enhanced array compares to a conventional hydrophone-based array.

4:25

2pSAb3. Wavenumber domain analysis of plates with embedded acoustic black holes. Philip Feurtado and Steve Conlon (Appl. Res. Lab., The Penn State Univ. PO Box 30, University Park, PA 16804, paf932@arl.psu.edu)

Acoustic black holes (ABHs) have been developed and demonstrated as effective, passive, lightweight bending wave absorbers that reduce the structural vibration and radiated sound power of plates. By introducing a gradual change in the local plate thickness, the bending wave speed is reduced and the transverse vibration amplitude is increased. Energy can then be effectively dissipated through material losses or attached damping treatments. In this paper, wavenumber spectra were generated from the vibrational responses of a uniform plate, an undamped ABH plate, and a damped ABH plate. The results showed that wavenumber transform analysis is a useful method for investigating and characterizing ABH performance and behavior. The results also demonstrated that ABHs can distribute supersonic bending waves into subsonic wavenumbers. The results will be useful for the design, characterization, and optimization of ABH systems for real structures

4:45

2pSAb4. Processing of defective ring data by "super-interpolators". Rudolph Martinez (Appl. Physical Sci., 49 Waltham St., Lexington, MA 02421, rmartinez@aphysci.com)

We develop an analytical-numerical scheme to make up for full or partial sensor failures in a ring array. The technique's goal is to restore missing information to a recorded signal in the form of a Hermitian covariance matrix for an arbitrary dependent variable. Our analysis breaks the problem down into two: first, by considering the special case of an imperfect array in which the positions and degrees of failure of the flawed sensors are known, and, second, by generalizing that approach to one in which neither piece of information is presumed nor in fact is the knowledge of how many sensors are affected required. An important application of the first part is the retrieval of the perfect signal at equispaced points that have been left un-instrumented for reasons of economy or lack of physical access (a missing sensor becomes equivalent to one with a known null gain). The problem's more general second part has led to a fundamental relationship for a periodic array's total number of sensors, the bandwidth of the ideal signal being restored, and the rank of an integral equation developed from one of Fourier series' dual statements of orthogonality. We call the products of our two-tier technique "super-interpolators" even though neither engages in that activity mathematically.

TUESDAY AFTERNOON, 24 MAY 2016 SALON J, 2:10 P.M. TO 5:25 P.M.

Session 2pSC

Speech Communication, Psychological and Physiological Acoustics, Architectural Acoustics, and ASA Committee on Standards: Intelligibility Challenges: Speakers, Listeners, and Situations

Amy T. Neel, Cochair
Dept. Speech and Hearing Sci., Univ. of New Mexico, MSC01 1195, Albuquerque, NM 87131

Suzanne E. Boyce, Cochair
Department of Comm. Sciences and Disorders, University of Cincinnati, Mail Location 379 University of Cincinnati, Cincinnati, OH 45267

Chair's Introduction—2:10

Invited Papers

2:15

2pSC1. The role of rhythm perception in recognition and learning of disordered speech. Stephanie A. Borrie (Dept. of Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84322, stephanie.borrie@usu.edu) and Kaitlin L. Lansford (School of Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL)

The musical advantage describes the idea that musicians, relative to non-musicians, are more successful at deciphering speech in challenging listening conditions. One assumption is that fine-tuned rhythm perception skills in the musical domain translate to fine-tuned processing of rhythm cues in the speech domain. But what happens when the rhythm cues afforded by the speech signal are, themselves, disordered? This study investigated whether the ability to perceive musical rhythms provides a perceptual advantage for recognition (initial intelligibility) and learning (intelligibility improvement) of dysarthric speech—a neurologically degraded acoustic signal characterized by rhythm abnormalities. Fifty young, normal hearing adults participated in two key tests including a rhythm perception test and a perception and learning test with standard pretest, training, and post-test phases. Initial intelligibility scores for each participant reflected words correct on the pretest, and intelligibility improvement scores reflected post-test words correct minus pretest words correct. The results revealed that rhythm perception scores predicted intelligibility improvement scores but not initial intelligibility scores. Findings are discussed in relation to theoretical models regarding the link between music and speech processing, and offer direction for new models that consider the perceptual consequence of rhythm abnormalities in disordered speech.

2:35

2pSC2. Speech intelligibility for native and non-native talkers and listeners. Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu)

Much of daily communication occurs under a range of suboptimal conditions related to talker, listener, and signal characteristics. These "adverse" conditions (Mattys *et al.*, 2012) increase intelligibility variation and impact perceptual processes, representations, attention, and memory functions. In this talk, I focus on two factors that condition intelligibility variation, namely, when interlocutors are non-native speakers of a target language and when communication occurs in noise. First, I will discuss a set of experiments examining how native and non-native talkers modify their speech with the goal of enhancing their intelligibility and the extent to which these modifications enhance word recognition in noise for native and non-native listeners. I also present results from studies examining how listeners adapt to more and less familiar foreign accents of varying intelligibility when presented in noise. These studies provide insights into the interplay between physical and mental factors in the production and perception of intelligible speech. They further reveal constraints on talker-listener adaptation processes during communication. A detailed understanding of speech intelligibility variation for native and non-native talkers and listeners will enhance our understanding of the compensatory and cognitive mechanisms that allow speech comprehension in naturalistic conditions.

2:55

2pSC3. Listener profiles for dysarthric and accented speech. Amy T. Neel (Dept. Speech and Hearing Sci., Univ. of New Mexico, MSC01 1195, Albuquerque, NM 87131, atneel@unm.edu)

In order to determine which aspects of speech contribute most to speech intelligibility, listeners rated several speech features for sentences produced by speakers with dysarthria and for passages read aloud by speakers of English as a second language. For the dysarthric speech, speech components included rate, stress, intonation, articulation, voice quality, and nasality in addition to speech intelligibility. For the accented English passages, the speech components were rate, stress, intonation, articulation, and other aspects of speech as well as overall intelligibility, accentedness, and language competence. For both types of speech, articulation was the best predictor of intelligibility ratings. Voice quality, rate, and nasality made minor contributions to intelligibility ratings for dysarthric speech. Rate also contributed slightly to intelligibility ratings for the accented English passages. Potential differences in the salience of speech components for listeners will be discussed. Use of the perceptual profile approach for planning treatment and assessing treatment efficacy for dysarthric and accented speech will also be addressed.

3:15

2pSC4. Applications of speech intelligibility assessment. Karen Payton (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747-2300, kpayton@umassd.edu)

There are many situations when it would be helpful to be able to predict how intelligible a talker will be to a specific listener in a specific environment. Metrics such as the Speech Intelligibility Index (SII) [e.g., Ricketts, Ear Hear. **17**, 124–132 (1996)] and the Speech Transmission Index (STI) [e.g., Humes *et al*., J. Speech Hear. Res. **29**, 447–462 (1986)] have been used for such purposes. This talk will review the history, philosophy, and effectiveness of these and other metrics in specific situations affecting the chain between talker and listener. Some of the situations include: hearing loss [e.g., HASPI: Kates and Arehart, Speech Commun. **65**, 75–93 (2014)], degradations due to room acoustics [e.g., SRMR: Santos *et al*., Speech Commun. **15**, 815–824 (2013)], and assistive communications technologies. The role of assumptions regarding speech in normal versus disorder populations will be briefly touched on.

3:35–3:50 Break

3:50

2pSC5. Industry testing of voice quality output from mobile devices. Carol Y. Espy-Wilson, Xinhui Zhou, and Ali Akrouf (OmniSpeech LLC, A.V. Williams Bldg., College Park, MD 20742, espy@umd.edu)

In general, the commercial evaluation of speech for telecommunication is focused on what is called “speech quality” rather than intelligibility per se. This talk will discuss the current state-of-the-art in the context of the objective test model 3QUEST (3-fold Quality Evaluation of Speech in Telecommunications) developed by Head Acoustics. 3Quest is an objective evaluation of transmitted speech with background noise and we will discuss the new ETSI standard that it is based on. The particular improvement relative to previous objective methods (PESQ and POLQUA) is that influence of different background noises is taken into account and the calculation of three mean opinion score (MOS) values, one each for the speech-only regions of the signal (SMOS), the noise-only regions of the signal (NMOS), and the total signal (GMOS). These scores allow for a more meaningful statement regarding the causes for the impression of quality. Using examples, we will discuss how signal noise, intelligibility, and quality are interrelated as signals are affected by various parameters.

4:10

2pSC6. Auditory and cognitive aging: Differences between speech intelligibility and speech-in-speech intelligibility. Margaret K. Pichora-Fuller (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, k.pichora.fuller@utoronto.ca)

Older adults, even those with normal or near-normal audiograms, report greater subjective listening difficulty in everyday life compared to younger adults. However, their difficulties are not well predicted from pure-tone or speech audiometry in quiet or noise. Speech intelligibility tests based on word recognition do not incorporate many of the sensory and cognitive challenges that listeners confront in everyday situations where it is necessary to understand a target talker when there are competing talkers in the auditory scene. Speech-in-speech intelligibility is affected by age-related differences in auditory and cognitive processing. Age-related differences in auditory temporal processing reduce access to periodicity and temporal envelope cues that serve stream segregation and spatial listening. Age-related differences in cognitive processing are reflected in difficulty remembering speech heard in multi-talker babble or switching spatial attention when there is uncertainty about the location of a target talker in a multi-talker display. Furthermore, when listening occurs in multitasking conditions (e.g., listening while walking), there are increased and competing demands for cognitive resources that may affect performance on listening and/or the competing tasks. These examples highlight the importance of both inter- and intra-individual differences in speech-in-speech intelligibility that depend on the interaction of auditory and cognitive processing abilities.

4:30

2pSC7. Production and perception of clear speech. Ann Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

When talkers are aware of listeners' speech perception difficulties due to hearing loss, noise, or a language barrier, they typically adopt an intelligibility-enhancing speaking style known as "clear speech." To the extent that clear speech is more intelligible than "conversational speech," a cross-style acoustic-phonetic comparison provides information about factors that affect speech intelligibility. I will present data from a project that tested the hypothesis that clear speech is guided by both universal, auditory-perceptual factors and language-specific, structural factors. The former serve to enhance the overall acoustic salience of the speech signal such that it is more resistant to the adverse effects of noise or listener-related perceptual deficits; the latter serves to enhance the realization of phonological contrasts. This hypothesis predicts that clear speech production shows predictable and systematic cross-language similarities and differences, and that the clear speech intelligibility benefit is modulated by the listeners' experience with the target language. To address these predictions, we present data from a cross-language comparison of clear speech production and a cross-population comparison of clear speech perception. These studies provide fundamental information about variability in speech intelligibility which may ultimately lead to effective, efficient, and listener-specific speech intelligibility enhancement strategies.

4:50

2pSC8. Lexical and nonlexical approaches to intelligibility in speech: An intermediate approach. Suzanne E. Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Mail Location 379 University of Cincinnati, Cincinnati, OH 45267, Suzanne.Boyce@uc.edu)

Intelligibility is a continuum with a floor and a ceiling. In this session, we are concerned with different ways of defining and quantifying intelligibility for different acoustic applications. A major break point in the continuum is defined by lexical transcription. For most applications, such as evaluating clinical and non-native speaker populations, the focus is on whether a reasonably adept listener would come up with a correct transcription under ideal conditions. For other applications, as in telecommunications or clear speaking style, we consider the effects of less-than-ideal conditions, cognitive fatigue, or listener pleasure. In general, however, the quantification of intelligibility is either defined by listener estimates, listener transcription or by broad statistical trends. In this paper, we consider an intermediate method of measuring the likely intelligibility of speech using acoustic characteristics specific to syllable shape, but not tied to transcription. Examples from measuring speaker fatigue, clear speaking style, and disordered source characteristics will be provided.

5:10–5:25 Panel Discussion

TUESDAY AFTERNOON, 24 MAY 2016 SOLITUDE, 1:00 P.M. TO 5:10 P.M.

Session 2pSP

Signal Processing in Acoustics: Comparison of Beamforming, Matched-Field Processing, and Time Reversal Techniques

Brian E. Anderson, Chair
N145 Esc, Brigham Young Univ., MS D446, Provo, UT 84602

Invited Papers

1:00

2pSP1. An overview of beamforming, matched-field processing, and time reversal techniques. William A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Marine Physical Lab., La Jolla, CA 92093-0238, wkuperman@ucsd.edu)

Beamforming (BF), matched-field processing (MFP), and time reversal (TR) are intimately related. On the simplest level, MFP is a generalization of BF in which the replicas are solutions to a wave equation descriptive of the environmental propagation physics as opposed to BF in which the replicas are typically plane waves. The TR process, on the other hand, is usually taken to be an active process, *independent of computer modeling,* in which the received signals at the elements of an array are time reversed and *physically* retransmitted. In the literature, one often finds examples for which the process of retransmission is implemented by a computer simulation. This is simply a linear version of BF or MFP. Further, passive TR does not refer to this latter case but is a special case. In all cases, there are adaptive processing augmentations. An interesting take on TR is that when TR works, it can be thought of as a kind of proof of existence that the medium supports the localization goal of BF and/or MFP. We review the connection between these various processors.

1:20

2pSP2. Out-of-band beamforming and matched field processing. David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133, drd@umich.edu), Brian M. Worthmann (Appl. Phys., Univ. of Michigan, Ann Arbor, MI), and Shima H. Abadi (Mech. Eng., Univ. of Washington – Bothell, Bothell, WA)

Beamforming and matched field processing (MFP) are array signal-processing techniques for estimating wave propagation directions and source locations, respectively. Both techniques can be formulated as spatial filtering operations that weight and combine the array recordings. In conventional plane- and spherical-wave beamforming, the weights are determined from analytical field models for the corresponding wave type. In conventional MFP, the weights are commonly determined from a computational field model that accounts for known propagation complications (reflection, refraction, scattering, diffraction, etc.) in the acoustic environment. Conventional MFP reduces to conventional beamforming when the environment is simple enough to be described by free-space propagation. Even at high signal-to-noise ratios, both techniques have limitations set by the recorded frequencies, the spacing of the array elements, the geometrical extent of the array, and mismatch between the recorded and modeled acoustic fields. Interestingly, these limitations can be overcome through signal processing techniques that recover out-of-the-signal band information from in-the-signal-band recordings. Here, in-band and out-of-band beamforming and MFP results are illustrated and compared using propagation simulations in free-space and multipath environments, and array recordings from a laboratory water tank and ocean propagation experiments. [Sponsored by the Office of Naval Research and the National Science Foundation.]

1:40

2pSP3. Particle-like wave packets in complex scattering systems. Benoît Gérardin, Jérôme Laurent (ESPCI ParisTech, PSL Res. Univ., CNRS, Université Paris Diderot, Sorbonne Paris Cité, Institut Langevin, 1 rue Jussieu, Paris 75005, France, benoit.gerardin@espci.fr), Philipp Ambichl (Inst. for Theor. Phys., Vienna Univ. of Technol. (TU Wien), Vienna, France), Claire Prada (ESPCI ParisTech, PSL Res. Univ., CNRS, Université Paris Diderot, Sorbonne Paris Cité, Institut Langevin, Paris, France), Stefan Rotter (Inst. for Theor. Phys., Vienna Univ. of Technol. (TU Wien), Vienna, Austria), and Alexandre Aubry (ESPCI ParisTech, PSL Res. Univ., CNRS, Université Paris Diderot, Sorbonne Paris Cité, Institut Langevin, Paris, France)

Waves propagating in complex media typically undergo diffraction and multiple scattering at all the inhomogeneities they encounter. As a consequence, a wave packet suffers from strong temporal and spatial dispersion while propagating through a scattering medium. This wave scattering is often seen as a nightmare in wave physics whether it be for focusing, imaging, or communication purposes. Controlling wave propagation through complex systems is thus of fundamental interest in many areas, ranging from optics or acoustics to medical imaging or telecommunications. Here, we study the propagation of elastic waves in a cavity and a disordered waveguide by means of laser interferometry. We demonstrate how the direct experimental access to the information stored in the scattering matrix of these systems allows us to selectively excite stationary scattering states and wave packets that follow particle-like bouncing patterns in transmission through (or in reflection from) a complex scattering landscape. Due to their limited dispersion, these particle-like scattering states will be crucially relevant for all applications involving selective wave focusing and efficient information transfer through complex media.

2:00

2pSP4. Multiple snapshot and multiple frequency compressive matched field processing. Kay L. Gemba, William S. Hodgkiss, and Peter Gerstoft (MPL/SIO, UCSD, Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu)

Matched field processing is a generalized beamforming method which matches received array data to a dictionary of replica vectors to locate and track a source. Its solution set generally is sparse since there are considerably fewer sources than replicas. The problem is also underdetermined since the number of sensors are less than the number of unique depth-range cells. Using compressive sensing (CS), the traditional spatial matched-filter problem is reformulated as a convex optimization problem subject to a row-sparsity constraint (RSC). The RSC selects the best match among the replica dictionary when using multiple snapshots. It is found that CS performance is equivalent to the Bartlett processor when comparing the sparse solution to the ambiguity surface's peak for any number of snapshots in a single frequency—single source scenario. Results also indicate that CS performs similarly to the adaptive white noise constraint processor in a multiple source scenario. The RSC can further be exploited to select a common depth-range cell from snapshots corresponding to multiple frequencies to improve the source tracking in the presence of data-replica mismatch. Results are demonstrated using both simulated and SWellEx-96 experiment data.

2:20

2pSP5. Localization of aeroacoustic sources by using time-reversal and beamforming techniques. David Marx, Vincent Valeau, Christian Prax, and Laurent-Emmanuel Brizzi (Institut PPRIME, UPR3346 Bat B17 Campus sud 6 rue marcel dorée, Poitiers 86022, France, laurent.brizzi@univ-poitiers.fr)

The study deals with the localization of aeroacoustic sources in flows by using array processing techniques. In such situations, the source is located in the flow and the propagation model must properly take into account flow effects, with possible temperature gradients. The resolution of the inverse problem can rely: (i) either on the numerical simulation of the propagation by using the time-reversal principle; (ii) or on using the beamforming technique, provided a high-frequency model of the Green function is used, such as that given by a simplified analytical model or by ray-tracing. The advantages and links between the two approaches are discussed. Some comparisons in terms of localization error are then provided, based on simulated data, both in the time and frequency domains. The case of a pulse in a shear flow is considered, showing similar performances for both methods. Then a harmonic source in a shear flow and in a jet flow is studied, with possible temperature gradients, showing the limitations of the beamforming method compared to the time-reversal-based method. Finally, examples of applications to experimental or numerical data are discussed.

2:40–2:55 Break

2:55

2pSP6. Acoustic beamfolding: New potentials enabled by interfacing reconfigurable origami and acoustic structures. Danielle T. Lynd and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., E540 Scott Lab, Columbus, OH 43210, lynd.47@osu.edu)

Acoustic transducer arrays yield large spatial and spectral change in energy transmission through strategic positioning of planar array elements. Yet, unless array elements are actively tuned through phase delay controls, the array performance characteristics remain fixed and suitable for limited purposes. Origami, the art of paper folding, is a means to introduce enormous topological change through simple, kinematic translation and rotation of connected, planar facets, which is one reason for its growing attention as a fluent vehicle to remarkably adapt system properties for multifunctional purposes. In this research, we integrate acoustic transducer array development and origami design principles to establish a new framework for adaptive acoustic energy shaping effected by simple geometric and kinematic folding relations. From a flat-folded strip to an unfolded plane, our theoretical and experimental results show that this transducer design concept leads to powerful means to tune the significance of radiated acoustic energy across orders of magnitude and to easily tailor the acoustic power transmission to the far field.

3:10

2pSP7. The effect of transducer directivity on properties of time reversal focusing. Miles Clemens, Matthew L. Willardson, and Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, theexperiment113@gmail.com)

This presentation explores the effect of transducer directivity on the focusing properties of time reversal acoustics. Time reversal is a signal processing technique that can be used to focus sound to a location in space for source reconstruction, communication, and intentional sound focusing. One loudspeaker and one microphone are used to produce a time reversal focus in a reverberation chamber. The primary axis of the loudspeaker is oriented in various different directions to produce a time reversal focus for each orientation. Since source directivity depends on the frequency of the sound emitted, these experiments are conducted for a low frequency band and a high frequency band such that these two frequency bands have fairly different directivities. Properties of these foci are compared to determine the overall effect of directivity on the time reversal focus.

3:25

2pSP8. Use of time reversal focusing to create a high amplitude focus of sound in air. Matthew L. Willardson, Miles Clemens, and Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, mwillardson@verizon.net)

Time reversal is a technique that may be used to achieve intentional sound focusing. The purpose of this presentation is to explore how intense the focus of airborne sound energy can be using time reversal. Experiments utilize loudspeakers and a single microphone placed in a reverberation chamber. Various frequency ranges and loudspeaker orientations are explored along with different types of loudspeakers in order to experimentally determine the optimum conditions for generating the loudest focus possible. The overall purpose of this study is to create an acoustic focus at nonlinear sound pressure levels in order to create a controllable source of nonlinear acoustic sound.

3:40

2pSP9. Comparison of inverse techniques for reproducing an extended, partially coherent sound field above a reflecting plane. Kevin M. Leete, Tracianne B. Neilsen, Blaine M. Harker, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, KML@byu.edu)

The relative abilities of two advanced array processing techniques to reconstruct a partially correlated noise source similar to a high-speed jet are compared. M-SONAH is a modification of statistically optimized near-field acoustical holography that includes multiple types of wavefunctions [A. T. Wall *et al*., J. Acoust. Soc. Am., **137**, 963–975 (2015)]. The hybrid method may be described as an advanced cross-beamforming algorithm with regularization [T. Padois *et al*., AIAA Paper 2013-2212]. A numerical, multiple wavepacket-based source distribution provides a reasonable model of the frequency-dependent directivity and coherence properties of a jet noise sound. An image source was included to account for a reflecting ground plane and the numerically modeled sound field was used an input to both the M-SONAH and hybrid methods. For M-SONAH, the image source was handled using an additional set of cylindrical wavefunctions and for the hybrid method, sources were assumed to be located both above and below the reflecting plane. While both methods provide accurate reconstructions, for this test case, the hybrid method performs better at higher frequencies using a realistic measurement aperture and density. [Work sponsored by the USAFRL SBIR program.]

3:55

2pSP10. Time reverse imaging of tsunami waveforms. Jakir Hossen, Phil R. Cummins (Res. School of Earth Sci., Australian National Univ., Canberra, ACT, Australia), and Jan Dettmer (Res. School of Earth Sci., Australian National Univ., 3800 Finnerty Rd., Victoria, Br. Columbia V8W 3P6, Canada, jand@uvic.ca)

We consider the application of Time Reverse Imaging (TRI) to tsunami waveform data, in order to recover the initial sea surface displacement associated with the tsunami source. We use as a case study the tsunami triggered by the March 11, 2011, Tohoku-Oki earthquake, for which an unprecedented number of high-quality observations are available. The method represents the tsunami source by dividing the source region into a regular grid of "point" sources. For each of these, a tsunami Green's function (GF) is computed using a basis function for sea surface displacement whose support is concentrated near the grid point. We apply the TRI method to estimate initial sea surface displacement at each source grid point by convolving GFs with time-reversed observed waveforms recorded near the source region. The result for initial sea surface displacement obtained via TRI agrees well with other models obtained using more traditional inversion methods. We show that the TRI method has potential for application in tsunami warning systems, as it is computationally efficient and can be used to estimate the initial source model using near-field data, and this TRI source model provides accurate and realistic predictions of far-field tsunami waveforms.

4:10

2pSP11. "Knocked Over!": A visual demonstration of time reversal focusing using bending waves in a thin plate. Sarah M. Young, Christopher Heaton, and Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, sarahmyoung24@gmail.com)

The purpose of this research is to develop a visual demonstration of time reversal focusing in a thin plate to be used as a teaching tool. This presentation will discuss the various plate materials tested to provide the optimal conditions for time reversal focusing. Specifically, the reverberation time in each plate and the sensitivity of the source of vibrations are quantified to provide the best spatially confined focus as well as the highest focal amplitude possible. A vibration speaker and a scanning laser Doppler vibrometer (SLDV) are used to provide the time reversal focusing. A series of objects are placed on the plate in an attempt to knock over objects near the focal location and not elsewhere. Spatial mapping of the plate vibration using the SLDV will be shown to illustrate why some objects fall and others do not.

4:25

2pSP12. On the alternative objective functions for minimum variance distortionless response technique in order to restore the original source. Yuling Tzeng and Gee-Pinn J. Too (Dept. of Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd. NCKU, Tainan, Taiwan, yulingtzeng@gmail.com)

Minimum Variance Distortionless Response (MVDR) technique is one of the signal processing ways for noise reduction. Its principle is to minimize the output power of the signal. This study simulated an underwater acoustic field using ray acoustic method. The acoustic field consists of a signal source, multiple noise sources and a line array hydrophones. The signal source is ship-radiated noise. The simulated combining signals received by the hydrophone array are used to restore the original signal source by MVDR technique. The study discusses the different objective functions given in MVDR in order to restore the original source. Then, the results are compared in computing time, effectiveness and adaptability. The results indicate that MVDR can be used to restore the original signal source with very high correlation coefficient up to 90% between the original signal and the restored one.

4:40

2pSP13. Application of time reversal mirror and minimum variance distortionless response technique for source localization. Gee-Pinn J. Too and Yuling Tzeng (Dept. of Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd. NCKU, Tainan 70101, Taiwan, toojames@yahoo.com)

Time difference of arrival (TDOA) and time reversal mirror (TRM) are commonly used in acoustic source localization. TDOA has excellent results in free field, but fair results in multiple reflections environment. TRM is applicable in multiple reflections environment and for multiple sources localization. The study continues the advantage of TRM and combines MVDR technique of noise reduction in order to find source locations. The results indicate that the combined MVDR technique gives better noise reduction than TRM in multiple reflections environment. From the drawing of energy diagram of acoustic waveguide, the combined MVDR technique will focus energy on signal source and it also reduces the noise energy in the other region. Comparing between TRM and the combined MVDR for source localization, the combined MVDR gives better resolution than TRM does. In addition to give better resolution, the combined MVDR will find the source location by finding the maximum energy location, on the other hand, TRM is unable to reduce reflection interference effectively which might lead to misjudge the source locations. However, the combined MVDR needs more computing time than TRM does.

4:55

2pSP14. Underwater acoustic communication in time-varying channel environment based on passive time reversal. Xiao Han, Jingwei Yin, Ge Yu, and Xiao Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Bldg., Nantong St., Nangang District, Harbin, Harbin 150001, China, hanxiao1322@126.com)

Channels estimated from probe signal or training symbols will mismatch with the true channels in passive time reversal (TR) underwater acoustic (UWA) communication when the ocean environment is time-varying, leading to poor channel equalization performance. In order to solve this problem, this paper studies Block-based TR processing structure. The received data are divided into small blocks and the channel estimated from previously decoded symbols is used to match filter the current block to suppress the channel mismatch. Decision feedback equalizer (DFE) is also combined with Block-based TR to remove the residual inter symbol interference (ISI) and Doppler effect (Block-based TR-DFE). Mobile UWA communication experiment was conducted near Xiao Changshan island, Dalian in July, 2015. Though the received data are preprocessed by resampling, residual Doppler effect still exists due to ununiform motion between the receiver and transmitter, making the channel time-varying fastly. Block-based TR realized effective tracking and compensation for time-varying channels. The output signal-to-noise ratio (SNR) after equalization is greatly improved and the bit error rate (BER) is significantly reduced.

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D. S. Houser, Chair ASC S3/SC 1
National Marine Mammal Foundation, 2240 Shelter Island Drive Suite 200, San Diego, CA 92106

K. Fristrup, Vice Chair ASC S12
National Park Service, Natural Sounds Program, 1201 Oakridge Dr., Suite 100, Fort Collins, CO 80525

Accredited Standards Committee S3/SC 1 on Animal 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. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note that these meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2016.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance, and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria, or aquariums; or free-ranging wild animals.

TUESDAY AFTERNOON, 24 MAY 2016 ALTA, 3:15 P.M. TO 4:30 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C. J. Struck, Chair ASC S3
CJS Labs, 57 States Street, San Francisco, CA 94114 1401

P. B. Nelson, Vice Chair ASC S3
Department of SLHS, University of Minnesota, 115 Shevlin, 164 Pillsbury Drive S.E., Minneapolis, MN 55455

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. 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 9:15 a.m. on Tuesday, 19 May 2015.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance, and comfort.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R. J. Peppin, Chair ASC S12
5012 Macon Road, Rockville, MD 20852

A. A. Scharine, Vice Chair ASC S12
U.S. Army Research Laboratory, Human Research & Engineering Directorate
ATTN: RDRL-HRG, Building 459, Mulberry Point Road, Aberdeen Proving Ground, MD 21005 5425

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. 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, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note that these meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 24 May 2016..

TUESDAY EVENING, 24 MAY 2016 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday are as follows:

Engineering Acoustics (4:30 p.m.)	Salon A
Acoustical Oceanography	Salon J
Animal Bioacoustics	Salon I
Architectural Acoustics	Snowbird/Brighton
Physical Acoustics	Salon H
Psychological and Physiological Acoustics	Salon B/C
Structural Acoustics and Vibration	Salon G

Session 3aAA

Architectural Acoustics: Relating Perception to Room Acoustics Measurements and Metrics in Performing Arts Venues I

Michelle C. Vigeant, Cochair
Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Gregory A. Miller, Cochair
Threshold Acoustics, LLC, 53 W. Jackson Boulevard, Suite 815, Chicago, IL 60604

Chair's Introduction—8:30

Invited Papers

8:35

3aAA1. Insights from three profiling studies of auralized concert hall acoustics: Support for a three dimensional perceptual space? Antti Kuusinen, Tapio Lokki, Jukka Pätynen, and Sakari Tervo (Comput. Sci., Aalto Univ., Otaniementie 17, Espoo FI-00076, Finland, antti.kuusinen@aalto.fi)

Since 2009, we have now completed three individual vocabulary profiling studies of auralized concert hall acoustics. In-line with the work of others, these studies have verified that the main perceptual aspects describing acoustic differences are, more or less, loudness, reverberance, width, envelopment, definition/clarity, brightness, bass, and proximity/intimacy. Our studies, however, do not indicate that these aspects would vary independently when moving from hall of to hall, but that the perceptual changes in many aspects generally co-occur. While independency can be understood in terms of attention, and the number of perceptual aspects by considering the different "tastes" of listeners, it seems that the perceptual differences between halls are actually governed by three main factors. The interpretation of these factors depends on the viewpoint and here we discuss some possible interpretations based on these three profiling experiments and the literature as well as the implications concerning the objective parameters.

8:55

3aAA2. Objective metrics, subjective descriptors, and their elasticity in the face of experience. Jonah Sacks (Acentech, 33 Moulton St., Cambridge, MA 02138, jsacks@acentech.com)

Acousticians have long associated certain objective metrics with subjective aural qualities: C80 with clarity, for example. But the subjective quality of sonic clarity is complex and nuanced, and the descriptor means different things to different people. C80 describes the strength of early reflections relative to overall reflected sound energy. But these reflections may either help or hinder perceived musical clarity in different halls, different conditions of use, and to different listeners. New research, listening experience, and developments in room acoustics design challenge us to update our understanding and use both of subjective aural qualities and descriptors (such as clarity), and of objective metrics (such as C80). This talk will cite research (Lokki *et al.*, Griesinger, etc.), project examples, and the author's own listening observations.

9:15

3aAA3. Multimodal perception in concert halls: Where do we look when we listen? Anne L. Minors (Sound Space Vision, Studio 2 Tay House, 23 Enterprise Way, London SW18 1FZ, United Kingdom, anne.minors@ampcstudio.com)

Concert hall design is at a crossroads between its origins, which have been unamplified orchestral music and singing, and the forces of popular music, which depend mostly on amplified sound and multimedia accompaniment. Concurrently, there has been a revolution in the way that the buildings are designed in the last 25 years. Computer modeling techniques enable architects and engineers to conceive and build complex geometrical forms and acoustic engineers to analyze future building interiors to promote a rich sound experience. However, despite a concert being a multisensory experience, relatively little work has been done on investigating how the visual aspects of the concert hall may impact on the acoustical experience. This paper investigates multisensory perception in different concert halls by examining where people look when actively listening to music and whether this affects how they perceive the experience.

9:35

3aAA4. How reproducible are psychoacoustic listening tests on spatial impression in auditoria? Ingo B. Witew, Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de), and Aleksandra Pieczykolan (Dept. of Psych., Univ. of Wuerzburg, Würzburg, Germany)

In three psychoacoustic experiments, it was shown that changes in frequency (pitch) and level (loudness) of stimuli lead to significant differences in the perception of apparent source width (ASW). Due to partially identical test conditions in two of the experiments, it is possible to analyze the collected data in regard to stability of ASW perception over repeated testing sessions and to study inter-individual differences. This analysis shows that intra-individual effects of stimulus loudness and pitch are consistent across participants, whereas the absolute level of ASW perception varies between individuals and between experimental sessions. Using this data, possible influences such as listener training effect over repetition of test sessions are discussed. The presented results provide insights into possible improvements of investigating ASW in future studies.

9:55–10:10 Break

10:10

3aAA5. Perception of spatial impression changes due to source movement. Sungbeen Cho and Lily Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, sungbeen@huskers.unl.edu)

Few investigations have studied how spatial impression at a receiver position in a performing arts venue varies as a source moves across the stage. This paper reviews an evaluation method for spatial impression by studying the relative change in received sound energy due to source movement, using a proposed metric called the Interaural Level Difference Correlation Range (ILD-CR). Previous work by the authors indicated that this metric is able to quantify varying spatial impression changes as a source moves across a stage, better than standard spatial impression metrics. A subjective study is now being conducted to assess perception of different ILD-CR in assorted spaces. Can listeners perceive different ILD-CR, and consequently differentiate between spatial impressions linked to different performing arts venues as source position changes? Results from the preliminary study are presented.

10:30

3aAA6. Relating listener envelopment to specific time segments in early and late portions of sound fields. David A. Dick and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dad325@psu.edu)

Listener envelopment (LEV) in concert halls is a function of both the spatial and temporal properties of the room impulse response (IR). This study used measured spatial IRs that were modified to study which segments of the early and late parts of the IRs contribute the most to the perception of LEV. Measurements were obtained in the Peter Kiewit Concert Hall in Omaha, NE, using an Eigenmike 32-element spherical microphone array, and processed for third-order Ambisonic reproduction over a 30-loudspeaker array. A subset of the IRs were identified with either an exceptionally high or low amount of LEV. These IRs were modified such that some time segments of the IRs were reproduced in full 3D (e.g., the early energy), while other segments were played only through a single loudspeaker in front of the listener (e.g., the late energy). Additional stimuli were generated that contain time portions of both the highly enveloping IRs and the unenveloping IRs. A subjective listening test was conducted in which listeners rated the LEV of the modified IRs convolved with anechoic music. Results will be presented that compare the LEV ratings to objective measurements of the sound fields. [Work supported by NSF Grant 1302741.]

10:50

3aAA7. Using individually equalized playback of binaural recordings from an electronic orchestra to determine the effects of early reflections on proximity, localization, and loudness. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt. Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Reflections in the first 50 ms are often assumed to enhance sound quality, but data supporting the assumption are lacking. We need to know when reflections are beneficial, and when they are not. Work by Lokki *et al.* may help to answer this question, but we have developed a binaural recording and playback technique that yields as good or better results. The key is individual equalization from the microphone to the eardrum of a listener. We have developed a quick and non-invasive method of headphone equalization that produces accurate timbre and frontal localization without head-tracking. This enables binaural impulse responses from a number of instrument positions in real halls to be manipulated by adding or deleting individual reflections and then evaluated with blind listening tests. Data from Boston Symphony show that early lateral reflections in the front of the hall are typically inaudible, but early medial and lateral reflections in the rear of the hall can reduce or eliminate proximity, clarity, impact, envelopment, and the ability to localize individual instruments. Deleting one or two of the earliest and strongest reflections improves all of those perceptions with no effect on loudness.

Session 3aAB

Animal Bioacoustics, Noise, and ASA Committee on Standards: Effects of Noise on Animals

John Hildebrand, Cochair
Scripps Institution of Oceanography, University of California San Diego, Mail Code 0205, La Jolla, CA 92093

David K. Mellinger, Cochair
Coop. Inst. for Marine Resources Studies, Oregon State University, 2030 SE Marine Science Dr., Newport, OR 97365

Chair's Introduction—8:15

Invited Papers

8:20

3aAB1. Characterizing ambient noise in marine and terrestrial settings. John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu)

Ambient noise results from both anthropogenic and natural sources. In a marine setting, low frequency ambient noise is dominated by anthropogenic sources: commercial shipping and seismic exploration. Marine ambient noise in the mid-frequency band is primarily due to sea surface agitation: breaking waves, spray, bubble formation and collapse, and rainfall. Various sonars (e.g., military and mapping), as well as small vessels, contribute anthropogenic noise at mid-frequencies. At high frequencies, acoustic attenuation becomes extreme so that all noise sources are confined to an area close to the receiver. In a terrestrial setting, ambient noise is often dominated by anthropogenic sources, from aircraft and traffic. Most terrestrial measurements of ambient noise are conducted at or near the ground surface, where temperature and wind gradients create complex sound propagation environments. Likewise, obstacles such as vegetation are important for sound propagation, as well as absorption and reflection from the ground surface. Flowing water and wind are natural sources of terrestrial ambient noise. The relationship between animal bioacoustics and ambient noise will be discussed, as a background against which studies of the impact of anthropogenic noise on animals are conducted.

8:40

3aAB2. Overview of the behavioral effects of noise on animals. David K. Mellinger (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu)

Animal responses to noise vary with both the noise itself and with the condition, and especially the behavioral state, of the animal. Responses include movement away from or toward the noise source on time scales from seconds (immediate response) to minutes (changes in breathing rates of marine mammals) to months or years (abandonment or adoption of habitats); changing the frequency, duration, intensity, or rate of occurrence of vocalizations; non-response due to masking of sounds important for mate-finding, predator detection, prey detection, navigation, socialization, parent-offspring bonds, etc.; increases in stress responses; and so on. Responses are mediated by a host of condition of the receiving animal, including its behavioral state (feeding, resting/sleeping, traveling, advertising for mates, etc.), its age/sex class, its mating status, its location within its habitat, past exposure to noise, presence of conspecifics, presence of predators, and many more factors. An overview of these topics is presented. [Work supported by ONR and LMR.]

9:00

3aAB3. Noise, national parks, and the wildlife therein. Kurt M. Fristrup, Megan F. McKenna (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov), and Rachel Buxton (Fish, Wildlife, and Conservation Biology, Colorado State Univ., Fort Collins, CO)

Monitoring at more than 600 sites in National Park Service (NPS) units has shown that noise poses widespread concerns: a contaminant to the physical environment, an infringement on superlative visitor experience, and a sensory burden for wildlife. NPS acoustical data were generalized into maps predicting sound levels for the coterminous U.S. These maps document the spatial scope and intensity of noise burdens on wildlife for park and other protected areas in the U.S. Although NPS units and other protected areas generally have sound levels a few decibels lower than adjacent unprotected land, 15% of all protected areas have sound levels 10 dB higher than predicted natural levels (median A-weighted levels). Designated critical habitat for 20 threatened animal species (US Fish and Wildlife Service) also exceed this 10 dB noise exposure criterion. Wildlife responses to noise have been documented in a wide range of taxa and habitats, with the past six years witnessing rapid growth in this research. Notably, controlled playback studies have been performed in otherwise pristine habitats to demonstrate that noise alone affects habitat utilization, foraging effectiveness, and breeding success. Noise may still present underestimated threats to wildlife; it certainly presents underutilized opportunities for habitat restoration.

9:20–9:35 Break

9:35

3aAB4. Insights into airplane overflight effects on bioacoustic activity levels from long-term acoustic monitoring. Susan Parks, Samuel L. Denes, Leanna P. Matthews (Biology, Syracuse Univ., 107 College Pl., Rm. 114, Syracuse, NY 13244, sparks@syr.edu), Pramod K. Varshney (Dept. of Elec. Eng. and Comput. Sci., Syracuse Univ., Syracuse, NY), and Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

The National Ecological Observatory Network (NEON) has established a framework through which a variety of environmental metrics will be continuously monitored measured for multiple decades at stations located across the United States. We describe a multiyear project that demonstrates the benefits of continuous acoustic monitoring at NEON sites. By utilizing long-term recordings, a wealth of data relating to species presence, animal behavior, and anthropogenic disturbance can be collected without the presence of human researchers. These recordings allow the measurement of contributions of anthropogenic activity to the cumulative sound energy at these locations and the response of all acoustically active species in the environment to individual events. Data from the NEON site at Harvard Forest, MA, were analyzed to assess the number of aircraft overflights detected at the location over the course of one year. The bioacoustic activity levels before, during and after were quantified for a subset of these events. Adding acoustics to the measurements already collected under NEON protocol can provide high-resolution information on the acoustic impact of human activities at these locations and allow for long-term monitoring with ground truth assessment of acoustic biodiversity. [Project supported by NSF award #1340669.]

9:50

3aAB5. Recent advances in scientific understanding of the effects of sound from seismic surveys. Robert Gisiner (Int.. Assoc. of Geophysical Contractors, 1225 North Loop West, Ste. 220, Houston, TX 77008, bob.gisiner@iagc.org), Jennifer Miksis-Olds (The Penn State Univ., State College, PA), and Sarah L. Tsoflias (Chevron North America E&P Co., Houston, TX)

The E&P Sound & Marine Life Joint Industry Programme (SML JIP), a partnership of 13 oil and gas companies and associations, funds independent scientific research to increase understanding of the potential effects of E&P sound on marine life. To advance understanding of the interaction between sound from oil and gas operations and marine life, the JIP identifies and commissions research to: (1) support planning of E&P projects and risk assessments, (2) provide the basis for appropriate operational measures that are protective of marine life, and (3) inform policy and regulatory development. SML JIP research categories include sound source characterization and propagation, physical and physiological effects and hearing, behavioral impacts and biological significant effects, and technologies for monitoring and mitigation. Highlights of projects to better characterize the source properties of seismic air sources, understand the hearing of Arctic pinnipeds, assess hearing recovery in marine mammals exposed to impulse sounds, and advance monitoring and mitigation technologies such as passive acoustic monitoring and alternative sound sources will be presented.

10:05

3aAB6. Relating the decreasing frequency of Sri Lankan pygmy blue whale calls to the local soundscape. Jennifer L. Miksis-Olds (Appl. Res. Lab, Penn State, PO Box 30, Mailstop 3510D, State College, PA 16804, jlm91@psu.edu) and Sharon Nieukirk (Oregon State Univ., Newport, OR)

Sri Lankan pygmy blue whale calls consist of three components: (1) low frequency pulsive unit, (2) frequency modulated upsweep, and (3) long tonal downsweep. The (~100 Hz) tonal downsweep is the most distinct of the call units and lasts 20–30 s. Spectral characteristics of the tonal downsweep and long-term patterns of environmental sound levels were analyzed from the Comprehensive Nuclear-Test-Ban Treaty International Monitoring Station at Diego Garcia in the Indian Ocean from 2002 to 2012. Average weekly spectral frequency peaks and ambient sound levels were computed. The peak frequency of Sri Lankan pygmy blue whale calls decreased from approximately 107 Hz to 100 Hz over a decade corresponding to a 0.55 Hz/year rate of decrease. To date, this is the largest rate of decrease observed for any blue whale call. Analysis of the ambient sound levels in the vocalization and adjacent bands did not exhibit equivalent patterns in source level trends. Potential drivers of the observed trends will be discussed. [Work supported by the Office of Naval Research.]

10:20

3aAB7. Construction noise impact on wild birds. Pasquale Bottalico (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Lansing, MI 48910, pb@msu.edu)

Almost all bird species use acoustic signals to communicate between conspecifics or recognize biological signals, to mate, to detect the sounds of predators and/or prey, to perform mate selection, to defend their territory, and to perform social activities. Noise generated from human activities (in particular by infrastructure and construction sites) has a strong impact on the physiology and behavior of birds. In this work, a quantitative method for evaluating the impact of noise on wild birds is proposed. The method combines the results of previous studies that considered the effect of noise on birds and involved noise mapping evaluations. A forecast noise simulation was used to generate maps of (1) masking-annoyance areas and (2) potential density variation. The results permit a localization of the areas with greater impacts on birds. The mitigation interventions should be focused on these areas in order to balance bird habitat conservation and human use of land. The forecast results should be interpreted by ornithologists and merged with information collected during the monitoring of the areas and with the habitat suitability maps.

Session 3aBA

Biomedical Acoustics: Controlled Drug Delivery and Release with Focused Ultrasound

Costas Arvanitis, Cochair
Radiology, Harvard Medical School, Brigham and Women's Hospital, 221 Longwood Avenue, Room 514a, Boston, MA 02115

Alexander Kilbanov, Cochair
Univ. of Virginia, Cardiovascular Div., CVRC, MR4RM3147, 409 Lane Rd., Charlottesville, VA 22908-1394

Invited Papers

8:15

3aBA1. Ultrasound-mediated drug targeting to tumors: Revision of paradigms through intravital imaging. Natalya Rapoport (BioEng., Univ. of Utah, 36 S. Wasatch Dr., Rm. 3100, Salt Lake City, UT 84112, natasha.rapoport@utah.edu)

In collaboration with Dr. Brian O'Neill (Houston Methodist Research Institute), the intravital fluorescence microscopy was performed using a customized Nikon A1R system to monitor the effect of ultrasound on the extravasation and tissue diffusion of various potential drug carriers including individual polymeric molecules, polymeric micelles, phase-shift nanoemulsions, and nanoemulsion-encapsulated drug. Carrier and drug extravasation and tissue accumulation was compared for the normal and tumor tissue upon intravenous injections to pancreatic tumor bearing mice. This approach allowed for the first time discriminating vascular and tissue compartments in the processes of the ultrasound-mediated drug delivery. Nanoemulsion accumulation in the tumor tissue was much faster than in the normal tissue. Without ultrasound, extravasation coefficient was threefold lower while tissue accumulation rate was two orders of magnitude lower for perfluorocarbon nanodroplets than for polymeric micelles. However, ultrasound application induced a 4.7-fold local enhancement of nanodroplet extravasation (to be compared with a 1.5-fold enhancement for micelles) and resulted in higher nanodroplet concentration and more uniform distribution in the tumor tissue. A kinetic model was suggested that allowed discriminating between various kinetic regimes of nanocarrier internalization in tumors of various sizes, cell density, and rigidity.

8:35

3aBA2. Triggered drug release from liposome-microbubble pendant complexes: From model systems to tumor therapy in a murine adenocarcinoma model. Alexander Kilbanov, Zhongmin Du, Galina Diakova (Cardiovascular Res. Ctr., Univ. of Virginia, Cardiovascular Div., CVRC, MR4RM3147, 409 Ln. Rd., Charlottesville, VA 22908-1394, alk6n@virginia.edu), Chien Ting Chin, William Shi, Balasundar Raju, and Ralf Seip (Philips Res. North America, Briarcliff Manor, NY)

Liposome-microbubble complexes allow delivery of drugs that cannot be otherwise associated with the bubble shell: water-soluble drugs and proteins. We prepared liposome-microbubble pendants by decorating biotinylated microbubbles with biotinylated liposomes via streptavidin. A model dye calcein was used as a model release marker. Using focused ultrasound (Philips TIPS, 1 MHz, 7 MPa) we were able to release ~30% of the entrapped dye. For an enzyme thrombin, ~11% of the entrapped material was released following pendant insonation. *In vivo* tumor therapy was performed with doxorubicin-liposome-microbubble pendants in a subcutaneous MC38 murine adenocarcinoma model. Doxorubicin was loaded in liposomes via an ammonium citrate gradient procedure. By using larger liposomes, >>80nm (as in Doxil/Lipodox), we prepared pendants carrying ~1 pg doxorubicin per particle. To avoid tumor blood flow stoppage caused by high-power insonation of microbubbles in tumor vasculature, we applied continuous sine wave ultrasound (Birtcher Megason, 1 MHz, 0.6 W/cm^2, 3 s on/10 s off, for 10 min) immediately following iv administration of doxorubicin-liposome-microbubble pendants (6mg/kg mouse body mass) under isoflurane anesthesia. Treatments were performed for two weeks, 2–3 times a week. Suppression of tumor growth in the experimental group was observed. All of control animals demonstrated rapid tumor growth. Overall, pendant structures may become new tools for ultrasound-triggered drug delivery. [Study supported in part via NIH R21/33 CA102880, EB016752.]

8:55

3aBA3. Low-frequency ultrasound for the delivery of therapeutics to the gastrointestinal tract. Carl M. Schoellhammer, Avi Schroeder, Ruby Maa (Chemical Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 76-661D, Cambridge, MA 02139, cschoell@mit.edu), Gregory Y. Lauwers (Massachusetts General Hospital, Boston, MA), Albert Swiston, Michael Zervas, Ross Barman, Angela M. DiCiccio (Chemical Eng., Massachusetts Inst. of Technol., Cambridge, MA), William R. Brugge (Massachusetts General Hospital, Boston, MA), Daniel G. Anderson, Daniel Blankschtein, Robert Langer, and Giovanni Traverso (Chemical Eng., Massachusetts Inst. of Technol., Cambridge, MA)

Rapid and effective drug delivery to the gastrointestinal (GI) tract can be a significant challenge. This is because of the harsh environment present in the GI tract and fast transit times in disease states. Physical enhancers, such as ultrasound (US), may enable the rapid delivery of therapeutics while circumventing the need for formulation development. Despite being investigated for other uses, low frequency US has not been studied for GI-based delivery previously. Our group has developed a hand-held device for the rapid delivery of

3a WED. AM

therapeutics to the colonic mucosa. The device utilizes low-frequency US, which is able to painlessly and reversibly permeabilize the tissue. Short, 1-minute treatments in 80 kg Yorkshire pigs were found to enhance the delivery of mesalamine, a drug used for the treatment of inflammatory bowel disease, 22.4-fold over a conventional enema. The safety and efficacy of US were further validated in a rodent colitis model. The delivery of proteins was also possible. US-mediated GI delivery has many potential applications ranging from localized treatment with anti-inflammatories to the more broad delivery of macromolecules. This new technology could prove invaluable in both clinical and research settings, enabling improved therapies and expansion of research techniques applied to the GI tract.

9:15

3aBA4. *In vitro* and *in vivo* platforms for focused ultrasound-controlled drug delivery and release. Costas Arvanitis (Radiology, Harvard Med. School, Brigham and Women's Hospital, 221 Longwood Ave., Rm. 514a, Boston, MA 02115, cda@bwh.harvard.edu)

Focused ultrasound (FUS) holds great promise for the development of effective and safe anticancer treatment protocols. FUS does not only allow to modulate the permeability of tumor vessels and enhance the extravasation of large molecules but to also locally trigger the release of their highly penetrating cargo. To utilize this potential *in vitro* and *in vivo,* platforms that allow for detailed assessment and optimization of the FUS-controlled drug delivery and release protocols are essential. We will present two such platforms for FUS-controlled drug delivery and release. First, we will introduce a novel acoustofluidic system with temperature and pressure sensor embedded that enables the accurate control of drug-release and the establishment of chemotherapeutic agent concentration gradients in a physiologically relevant 3D tumor model. Next, we will present an integrated US- and MR-guided FUS system for controlling the mechanical and thermal effects of FUS *in vivo*. Fast methods to visualize and control microbubble oscillations along with real time FUS-induced mild hyperthermia with this system will be shown in brain tumors of rodents. The FUS-induced release of the chemotherapeutic agent doxorubicin from a liposomal carrier is used to demonstrate the utility of the two systems for developing and optimizing new therapeutic protocols.

9:35

3aBA5. MRI-targeted delivery of brain-penetrating non-viral gene nanoparticles across the blood–brain barrier with focused ultrasound: Neurodegenerative disease application. Richard J. Price (Biomedical Eng., Univ. of Virginia, Box 800759, Health System, Charlottesville, VA 22908, rprice@virginia.edu)

The delivery of systemically administered nanoparticles to the brain is impeded by both the blood–brain barrier (BBB) and the nanoporous electrostatically charged extracellular matrix. However, we have previously shown that these barriers may be overcome by opening the BBB with MRI-guided focused ultrasound (FUS) and microbubbles (MBs) and engineering the nanoparticles to have "brain penetrating" properties via the addition of a dense PEG corona. Here, we first delivered non-viral reporter gene-bearing brain-penetrating nanoparticles (BPN) to rat brain using FUS and MBs, resulting in robust dose-dependent gene expression in the FUS-targeted region through day 28 and a transfection efficiency $>40\%$. Neurons and astrocytes were transfected equally, and neither toxicity nor gliosis were evident. We then tested whether the approach had therapeutic potential for treating Parkinson's disease by delivering neurotrophic (GDNF) gene BPN to the striatum of 6-OHDA treated (i.e., Parkinson's) rats. For GDNF BPN treated 6-OHDA rats, motor impairment tests (apomorphine-induced rotation and cylinder) revealed significant improvement and dopaminergic neuron density was fully restored in both striatum and substantia nigra pars compacta. We conclude that MRI-guided BPN delivery with FUS and MBs is a safe and effective strategy for brain transfection that has potential as a non-invasive treatment for Parkinson's neurodegeneration.

9:55–10:10 Break

10:10

3aBA6. Ultrasound- and microbubble-enhanced chemotherapy for treating pancreatic cancer: A phase I clinical trial. Spiros Kotopoulis, Georg Dimcevski (National Ctr. for Ultrasound in Gastroenterology, Haukeland Univ. Hospital, Jonas Lies vei 65, Bergen 5021, Norway, spiros.kotopoulis@gmail.com), Emmet Mc Cormack (Dept. of Clinical Sci., Univ. of Bergen, Bergen, Norway), Michiel Postema (Dept. of Phys. and Technol., Univ. of Bergen, Bergen, Norway), Bjorn Tore Gjertsen (Dept. of Internal Medicine,Haematology Section, Haukeland Univ. Hospital, Bergen, Norway), and Odd Helge Gilja (National Ctr. for Ultrasound in Gastroenterology, Haukeland Univ. Hospital, Bergen, Norway)

Experimental research of ultrasound to induce or improve delivery has snowballed in the past decade. In our work, we investigate the use of low-intensity ultrasound in combination with clinically approved microbubbles to enhance the therapeutic efficacy of chemotherapy. Ten voluntary patients with locally advanced or metastatic pancreatic adenocarcinoma were consecutively recruited. Following standard chemotherapy protocol (intravenous infusion of gemcitabine over 30 min), a clinical ultrasound scanner was targeted at the largest slice of the tumour using modified non-linear contrast imaging settings (1.9 MHz center frequency, 0.27 MPa peak-negative pressure), and SonoVue® was injected intravenously. Ultrasound and microbubble treatment duration was 31.5 min. The combined therapy did not induce any additional toxicity or increase side effect frequency when compared to chemotherapy alone. Combination treated patients were able to tolerate an increased amount treatment cycles when compare historical controls ($n = 63$); average of 8.3 ± 6.0 cycles, versus 13.8 ± 5.6 cycles. The median survival also increased from 7.0 months to 17.6 months ($p = 0.0044$). In addition, five patients showed a primary tumor diameter decrease. Combined treatment of ultrasound, microbubbles, and gemcitabine does not increase side effects and may have the potential to increase the therapeutic efficacy of chemotherapy in patients with pancreatic adenocarcinoma.

10:30

3aBA7. Ultrasound-mediated transport of nanoparticles and the influence of particle density. Harriet Lea-Banks, Eleanor Stride, and Constantin C. Coussios (Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, harriet.lea-banks@eng.ox.ac.uk)

A significant barrier to successful drug delivery in cancer therapy is the limited penetration of nanoscale therapeutics deep into tumors. Ultrasound mediated cavitation has been shown to promote nanoparticle transport, but achieving tumor wide distribution still represents a considerable challenge. The current study investigates the way in which nanoparticle drug-carriers may be designed to enhance penetration under ultrasound exposure. A computational model has been developed to predict the transport of a nanoparticle in an ultrasonic field in the presence of an oscillating microbubble, by a combination of primary and secondary acoustic radiation forces, acoustic streaming, and microstreaming. Experimental investigations were also performed in a tissue-mimicking phantom to study the transport of different types of particle, in the presence or absence of a microbubble ultrasound contrast agent, at ultrasound frequencies of 0.5 MHz and 1.6 MHz with peak pressures in the range of 0–2.0 MPa. Micro- and nanoparticles with contrasting density cores ranging from 1.0 g/cm^3 to 19.3 g/cm^3 were used for the study. Both the theoretical and experimental results showed that the denser particles exhibit significantly greater ultrasound-mediated transport than their lower density counterparts, indicating that this is a key consideration in the design of nanoscale therapeutics.

10:45

3aBA8. Ultrasound-mediated delivery of gadolinium and fluorescent-labeled liposomes through the blood-brain barrier. Muna Aryal (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., Boston, MA 02115, muna@bwh.harvard.edu), Jason Papademetriou (Biomedical Eng., Boston Univ., Boston, MA), Yong-Zhi Zhang, Chanikarn Power, Nathan McDannold (Radiology, Brigham and Women's Hospital, Boston, MA), and Tyrone Porter (Biomedical Eng., Boston Univ., Boston, MA)

The main objective of this study was to examine liposome extravasation across the BBB as a function of size after disruption via ultrasound and microbubbles. The liposomes were labeled with gadolinium (Gd) and fluorophore, thus enabling detection of extravasated liposomes via MRI *in vivo* and fluorescence methods in tissue, respectively. Liposomes labeled with gadolinium and fluorophore were prepared using lipid film hydration and extrusion to two different sizes (~80 nm and ~140 nm). Animals were divided into two groups based on the use of particle sizes. FUS–BBB disruption was produced in one hemisphere in 10 mice. Particles were injected before sonication. Sonications (0.69 MHz at 0.68 MPa) were performed in two locations combine with Definity (10 μl/kg). Acoustic emissions were recorded during FUS. T1 & T2*-weighted MRI were used to confirm Gd leakage and damage detection respectively. Mice were euthanized 5–24 h after FUS and post-process for fluorescence measurement. In sonicated area, Gd-leakage was detected in both groups at 5–24 after FUS but not on non-sonicated area. Fluorescence measurements from brain tissue homogenates suggest enhanced accumulation of liposomes in FUS versus non-FUS brain regions. These findings reveal that liposomes were able to extravasate via FUS-BBB disruption.

11:00

3aBA9. Thrombolytic efficacy of echogenic liposomes that co-encapsulate rt-PA and octafluoropropane gas. Himanshu Shekhar (Dept. of Internal Medicine, Univ. of Cincinnati, 3933 Cardiovascular Ctr.,, 231 Albert Sabin Way, Cincinnati, OH 45267, himanshu.shekhar@uc.edu), Shenwen Huang (Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Tao Peng, Melvin E. Klegerman, Shao- L. Huang, David D. McPherson (Dept. of Internal Medicine, Univ. of Texas Heath Sci. Center-Houston, Houston, TX), and Christy K. Holland (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Echogenic liposomes loaded with recombinant tissue-type plasminogen activator, rt-PA (tELIP), are under development for treatment of thrombo-occlusive disease. However, tELIP filled with air nucleate low stable cavitation activity when exposed to 120-kHz ultrasound, which limits the thrombolytic efficacy. We hypothesize that encapsulating octafluoropropane (OFP) gas in ELIP (OFP-tELIP) will enhance ultrasound-mediated stable cavitation activity and thrombolytic efficacy. A spectrophotometric method was used to assess the enzymatic activity of the rt-PA associated with OFP-tELIP. An *in vitro* flow model equipped with a time-lapse microscopy system was employed to observe human whole blood clots exposed to fresh-frozen plasma, rt-PA (0, 0.32, 1.58, and 3.15 μg/mL), and OFP-tELIP with and without ultrasound (120 kHz, 0.44 MPa peak-to-peak pressure). Ultraharmonic emissions indicating stable cavitation were measured using a passive cavitation detector. Sustained ultraharmonic activity was nucleated from OFP-tELIP when exposed to ultrasound, resulting in enhanced thrombolytic efficacy at an rt-PA concentration of 1.58 μg/mL. The results of this study demonstrate the advantages of encapsulating OFP within tELIP for use as a sonothrombolysis agent.

11:15

3aBA10. Shape mode stability of lipid-coated ultrasound contrast agent microbubbles. John T. Brlansky and Michael L. Calvisi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, mcalvisi@uccs.edu)

Ultrasound contrast agents (UCAs) are shell encapsulated microbubbles developed originally for ultrasound imaging enhancement. More recently, UCAs are being exploited for therapeutic applications such as drug and gene delivery. Ultrasound transducer pulses can induce spherical (radial) UCA oscillations, translation, and nonspherical shape oscillations, the latter of which can lead to breakup. Breakup can facilitate drug or gene delivery, but should be minimized for imaging purposes to increase residence time and maximize diagnostic effect. Therefore, an understanding of the interplay between the acoustic driving and shape mode stability of UCAs is important for both diagnostic and therapeutic applications. The present work couples a radial model of a lipid-coated microbubble with a model for bubble translation and nonspherical shape oscillation to predict shape mode stability for ultrasound driving frequencies and pressure amplitudes of clinical interest. In addition, calculations of the stability of individual shape modes, residence time, maximum radius, and translation are provided with respect to acoustic driving parameters and compared to an unshelled bubble. The effects of shell elasticity, shell viscosity, and initial radius on stability are investigated. The results show greater stability at higher values of shell elasticity and viscosity and at smaller radius, and provide guidance for optimizing shell design and ultrasound driving parameters with respect to shape stability.

Session 3aED

Education in Acoustics: Education in Acoustics: Student Posters

Murray S. Korman, Chair
Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402

All posters will be on display and all authors will be at their posters from 8:00 a.m. to 10:00 a.m.

Contributed Papers

3aED1. Nonlinear vibration experiment: Clamped circular elastic plate with granular material loading. Emily V. Santos (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

Experiments using a soil-plate-oscillator (SPO) involve a vertical cylindrical column of granular medium (masonry sand, glass spheres, uncooked brown rice, un-popped popcorn kernels, or even "Toasty Oats" ™ cereal) that is supported by an air-backed thin circular elastic acrylic plate (20.3 cm diameter and 3.2 mm thick) that is rigidly clamped to the bottom of a thick-walled aluminum tube. The soil column is driven from below using an electrodynamic system. Here, an AC coil placed on axis and below the plate, drives a 1 cm diameter 1.5 cm long rare earth magnet that is fastened to the underside center of the plate. The coil is electrically driven by an amplified swept sinusoidal slowly varying chirp. A small accelerometer attached to the magnet is used to measure the vibration. In nonlinear tuning curve experiments the resonant frequency decreases significantly with increased amplitude—representing a softening in the nonlinear system. For fixed amplitude the resonant frequency vs. the granular medium mass loading (over the plate) reaches a minimum and then increases with increased loading due to the granular medium's flexural stiffness—which overcomes the mass loading effects. For water loading, the frequency always decreases since there is no bending stiffness.

3aED2. Vibration experiments using a clamped circular elastic plate with edible granular material loading. Blair E. Lewis, Ebonie Smith, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu)

An apparatus called the soil-plate-oscillator (SPO), designed to study flexural vibration of a soil loaded plate, consists of a thin circular elastic (acrylic) plate (8 in. diam, 1/8 in. thick) clamped below a thick-wall cylindrical aluminum tube supporting a vertical soil (or sand) column or other granular material. A small accelerometer attached to a 1 cm diam rare earth magnet (which is fastened to the center of the plate from below) is used to detect the plate vibration response. The plate is driven from below by an AC coil (located coaxially below the plate and securely fastened) using an amplified swept sinusoidal current. The charge amplified accelerometer signal is measured versus frequency by a spectrum analyzer. With interest in studying light density granular media (with various grain sizes) experiments were performed with uncooked brown rice, quick oats, un-popped popcorn kernels pretzel gold fish crackers, and pretzel nuggets. The resonant frequency reaches a minimum and then increases with increased granular medium loading due to the material's flexural stiffness which overcomes the mass loading effects. Results (normalized to the unloaded frequency and clamped plate mass) are compared with dry sifted masonry sand. A theoretical model is used to help describe the effects.

3aED3. A soil-plate-oscillator apparatus for research projects and student demonstrations. Melissa Pineda Brown, Brianna D. Taliaferro, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu)

In studying the complex vibration of compliant buried objects in soils, a simplified model apparatus called the "soil-plate-oscillator" (SPO) has been useful in understanding resonant behavior. The SPO is an open column of granular medium supported at the bottom by a thin circular clamped elastic plate. A rigid vertical circular sleeve (sidewall) keeps the soil in a circular column. Two 4.5 in. I.D. plastic PVC closet flanges are used to clamp a 1/8 in. thick acrylic plate. The elastic plate is driven from below by an AC coil located coaxially below a rare earth magnet (fastened to the plate's underside at the center). An amplified swept sinusoidal chirp drives the coil. A small accelerometer, attached to the magnet, generates a charge amplified signal fed into a spectrum analyzer placed in the swept sine mode. Results for (a) dry sifted masonry sand and (b) 6 mm glass spheres are compared. In both experiments, the resonant frequency versus granular mass loading first decreases (reaching a minimum) and then increases with further loading due to the granular flexural stiffness which overcomes the mass loading effects. [R. A. Guyer and P. A. Johnson, *Nonlinear Mesoscopic Elasticity*, Wiley, 2009] discusses spherical granular media in Hertzian contact.

3aED4. Modeling acoustic landmine detection using a soil-plate oscillator. Miahanna K. Nguyen, Joshua M. Lewis, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu)

In laboratory acoustic landmine detection experiments a plastic cylindrical drum-like simulant is buried in a soil (or sand) tank. Airborne sound, generated from two subwoofer loudspeakers (located above the soil), drive the soil particles and subsequent particle vibration over the compliant top plate of the simulant. Measurements of tuning curve soil surface vibration particle velocity versus frequency are recorded for various scan locations across the soil surface in an effort to profile the buried simulant. Measurements of resonances "off the target" are also included in this study. The results can be modeled using a soil-plate-oscillator (SPO) apparatus. The SPO involves a vertical thick-wall cylindrical column of granular medium (sand, soil, pebbles, light density edible granular material like wheat germ or even uncooked brown rice) that is supported by a thin circular elastic (acrylic) plate (8 in. diameter, 1/8 in. thick) that is rigidly clamped at the bottom of the column. A small accelerometer placed on the granular surface (or a laser Doppler vibrometer) measures tuning curve results across the surface using a sweep spectrum analyzer. Profiles are compared for both the SPO and the simulant in an effort to model the results in the later—more complicated case.

3aED5. Ultrasonic scattering at 25 kilohertz in air by a hexagonal array of slender long rigid rods arranged to form a porous cylindrical target. Joshua R. Gong, Adela S. Rivera, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu)

Experiments were performed in the USNA Acoustics Lab using 30 cycle transient-bursts at 25.8 kHz in air (c = 340 m/s, $\rho = 1.2$ kg/m^3) to measure the backscattering of the ultrasonic pulse from a hexagonal array of stainless steel rods (diam d = 0.089 cm, length l = 91 cm). The lattice parameter was a = 0.22 cm. To construct the array, the rods were placed within the holes of two 30.5 cm square thin perforated aluminum sheets suspended 88 cm apart. A third non-perforated sheet hung 1 cm below prevented the rods from falling through. The diameter of the cluster was D = 10 cm. The filling fraction of the lattice was f = 0.15.The transmitting array (located 1.5 m from the cluster) was an eight-element discrete ring of diam 5.5 cm that produced an interference null at +/− 10 degrees and a −10 dB down minor lobe at +/− 17 degrees. A single 1 cm diam receiver element was located at the center of the ring. In a strip-map configuration, the transmitted beam translated 60 cm across the target recording echoes at 1 cm increments. The target's effective speed and density were 0.93c and 1.35ρ, respectively. See D. Torrent *et. al*. [Phys. Rev. Lett., **96**, 204302 (2006)].

WEDNESDAY MORNING, 25 MAY 2016 SOLITUDE, 9:00 A.M. TO 11:45 A.M.

Session 3aMU

Musical Acoustics and Education in Acoustics: Teaching Musical Acoustics Courses and Laboratories at any Level

Jack Dostal, Cochair
Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, NC 27109

Martin S. Lawless, Cochair
Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Chair's Introduction—9:00

Invited Papers

9:05

3aMU1. Teaching the descriptive physics of string instruments at the undergraduate level. Brian E. Anderson, Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., BYU, N283 ESC, Provo, UT 84602, bea@byu.edu)

At Brigham Young University, a general education course introduces students to the basic descriptive acoustic principles of music, speech, and audio. A third of this course focuses on the physics of musical instrument families. Three of these families include bowed string, plucked string, and struck string instruments. The concepts of driven systems and freely vibrating systems are taught, including the consequences of these excitation conditions. A hands-on lab for the course enables students to explore how the length, density, and tension of the string change the fundamental frequency. An in-class demonstration highlights the role of inharmonicity on the partial frequency values for these string instrument families. The combination of hands-on activity and demonstration aids the students in comprehending the basic acoustic principles behind string instruments and the reason for varying levels of inharmonicity between these instrument families.

9:25

3aMU2. A case study approach involving string harmonics to reinforce concepts of standing waves, superposition, and musical intervals... as well as the scientific process. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

A combined classroom/lab activity is described that employs a modified Interrupted Case Study approach to reinforce concepts of standing waves, superposition, and musical intervals. Students work cooperatively, with guidance from the instructor, to investigate the behavior of a plucked string. The activity is structured as a sequence of tasks that encapsulate the scientific process, requiring the application of critical thinking skills and knowledge from earlier classes. Students working in small groups propose a hypothesis about the behavior of a string that is plucked, drawing upon previous studies of superposition and standing wave behavior in a driven string. After a class discussion, one hypothesis is chosen as the basis for further inquiry. Each group then formulates specific testable predictions, with the constraint that only a mounted string (sonometer), finger, and ears be used in any experiment. After discussion, the class chooses a subset of these predictions to test. Groups formulate a detailed procedure (generally involving creating harmonics on the string), carry out the experiment, and interpret their results. Conclusions are shared among the class.

9:45

3aMU3. Things I learned while teaching a graduate level course on the acoustics of musical instruments. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

In spring 2013, I taught a graduate level course on the acoustics of musical instruments. A prerequisite of two courses on the fundamentals of acoustics and vibration allowed us to explore details and mathematical complexities. Due to time constraints and the decision to delve deep into details rather than provide a broad overview, the course focused on wind and stringed instruments. Both types of instruments were explored through the generator-resonator-radiator paradigm. We discussed nonlinear generators for winds (lip reed, mechanical reed, and air-reed) and realistic initial conditions for strings (finger plucked, struck by a nonlinear hammer, and the stick-slip bowing action); realistic resonators for winds (input impedance and the effects of holes, horns, and viscous losses) and strings (realistic boundary conditions and coupled motion); and radiation (bells and soundboards). This paper will describe some of the Mathematica animations used to illustrate the effect of tone holes on the input impedance of a woodwind instrument, the effect of mouthpiece volume on the input impedance of brass instruments, and the motion of plucked, struck, and bowed strings. We will discuss the challenges of teaching a graduate level course, and what I would do differently the next time I teach it.

10:05

3aMU4. Explaining microphones and loudspeakers in a musical acoustics course for non-scientists. Robert C. Maher (Elec. & Comput. Eng., Montana State Univ., 610 Cobleigh Hall, PO Box 173780, Bozeman, MT 59717-3780, rob.maher@montana.edu)

Contemporary courses in the field of musical acoustics rely upon basic audio engineering components such as acoustical transducers: microphones, accelerometers, and loudspeakers. Most students are very aware of the practical use and behavior of such devices, but they seldom have a useful understanding of the physical and engineering principles behind the design of these essential components. This paper presents a lesson example explaining microphones and loudspeakers for a general-interest undergraduate college course entitled “The Science of Sound.” The lesson includes a descriptive lecture component, examples and discussion, and a hands-on demonstration.

10:25–10:40 Break

10:40

3aMU5. Developing and teaching an interdisciplinary musical instrument design course at the Cooper Union. Martin S. Lawless (Dept. of Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, 201 Appl. Sci. Bldg., University Park, Pennsylvania 16802, msl224@psu.edu), Melody Baglione, and George W. Sidebotham (Dept. of Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, New York, NY)

Design is a major component of every student’s educational experience at the Cooper Union, a small university in New York City with architecture, art, and engineering schools, and yet, design courses including students from all three schools are uncommon. In the 2012–2013 academic year, a Musical Instrument Design course was proposed to encourage the collaboration of architecture, art, and engineering students. Musical acoustics appealed to these three types of students since the subject inherently combines aesthetic and physical principles. The course intended to accommodate the students’ wide range of music and physics backgrounds with lectures on music theory, acoustics and vibration, and the design of traditional instruments. Live demonstrations attempted to inspire hands-on learning, and in-class presentations facilitated the discussion of the diverse perspectives of the class. The course also included an outing to the machine shop where the class collaborated on making PVC flutes for each student. For a final project, the students were required to design and create a musical instrument. The Musical Instrument Design class was offered for zero credits in the Spring 2013 semester to gauge overall interest. Fourteen students enrolled in the course and maintained a good attendance throughout the semester.

Contributed Papers

11:00

3aMU6. Students and power tools: Lessons learned from building recorders in a Physics of Music class. Jack Dostal (Dept. of Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

In my divisional (general education) course “The Physics of Music,” my students each design and build a simple recorder using PVC pipe. The process is borrowed from “Flute Design” and “Flute Construction” labs by Peter Hoekje at Baldwin-Wallace University, as well as Pete Kosel’s flute hole calculator. The lab has been modified slightly to fit the constraints of our class, and is done over the course of two lab periods. We have tried several variations on the process and made plenty of mistakes. Typically, our students have little to no experience using power tools or shop equipment. Consequently, this tends to be one of the most adventurous labs of the term. In this talk, I describe some of the practical things I’ve learned and mistakes I’ve made while watching and helping my students build recorders. Among these are difficulties in creating fipples and knife edges, adventures in hole-drilling, finding reasonable finger hole positions, and student reaction to the finished product.

11:15

3aMU7. Engaging musical acoustics instructors beyond the members of the Acoustical Society of America. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

The members of the Technical Committee on Musical Acoustics (TCMU) of the Acoustical Society of America (ASA) have demonstrated their commitment to education in acoustics regularly by organizing and cosponsoring numerous education-related sessions at ASA meetings. Musical acoustics as a topic for general education science courses is a popular offering at many colleges and universities. A brief survey of the musical acoustics course offerings at various colleges is presented. Many of these courses are taught by instructors who have not interacted with TCMU members or attended ASA meetings. It is possible that the musical acoustics community is missing out on opportunities to interact with a population of educators who share our interests. Our community should be looking for ways to engage the instructors of musical acoustics courses in dialogue which is mutually beneficial; members of the TCMU may find that they will discover new ideas for teaching musical acoustics from beyond the ASA membership.

11:30

3aMU8. Electroacoustics laboratory assignments for computer music students. Edgar Berdahl and Stephen D. Beck (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

A series of pedagogical exercises are described that integrate concepts from traditional musical acoustics laboratory assignments with the fundamentals of computer music. Students are asked to rapidly prototype hybrid devices that contain both musical acoustic components as well as sensors, embedded audio signal processing, and loudspeaker drivers. To help students rapidly complete working prototypes, students are provided with working audio effects and sound synthesizer programs. Students learn the programming language (Pure Data) only by example for making small changes to the previously designed programs, as they are integrated with custom-made, physical acoustics components (e.g., D-I-Y vibrating strings, drums, columns of air, rattles, etc.). Music students first learned that building quality acoustic instrument components is harder than they originally thought. Although out-of-the-box thinking is regarded as an important factor in devising novel electroacoustic prototypes, students starting from previously existing designs and making small changes tended to be more successful (at least initially) than students striking out on completely new paths. Several pictures of completed instruments will be shown, and one or two example prototypes of electroacoustic instruments and electroacoustic audio effects will be presented live. The example "starter" programs in Pure Data will be made available for community members wishing to run them on the Raspberry Pi 2, as supported by the author's Satellite CCRMA platform.

WEDNESDAY MORNING, 25 MAY 2016 SALON J, 8:30 A.M. TO 11:25 A.M.

Session 3aPA

Physical Acoustics: Atmospheric Acoustic Phenomena I

John Paul R. Abbott, Cochair
Department of Physics and Astronomy, National Center for Physical Acoustics, University of Mississippi, 1 Coliseum Dr., Room 1044, Oxford, MS 38677

Gregory Lyons, Cochair
Aeroacoustics, National Center for Physical Acoustics, University, MS 38677-1848

Invited Papers

8:30

3aPA1. Accuracy requirements for unmanned aerial vehicle-based acoustic atmospheric tomography. Anthony Finn and Kevin Rogers (Defence & Systems Inst., Univ. of South Australia, Bldg. W, Mawson Lakes, Mawson Lakes, SA 5095, Australia, anthony.finn@unisa.edu.au)

If the acoustic signature is measured onboard a propeller-driven unmanned aerial vehicle and by an array of microphones on the ground, comparison between the projected and observed signals allows the propagation delays—and hence effective sound speed values of the intervening medium—to be determined. As the ray paths along which energy travels intersect within the atmosphere at multiple locations and angles, temperature and wind velocity fields may be estimated using tomography. This paper describes signal processing strategies for determining propagation delays at sub-millisecond levels of precision; the inverse problem and a model that allows meteorological observations to be incorporated into the tomographic solution; and the algebraic iterative reconstruction technique used to overcome the ill-posed nature of the inversion. A weakly sheared daytime convective atmospheric boundary layer—synthesised through use of large eddy simulation code that utilises pseudo-spectral differencing and solves an elliptic pressure equation—is used to show sub-millisecond levels of observation noise permits faithful reconstruction of target atmospheres. Particular attention is paid to the atmospheric layer below 100 m as this region typically experiences the greatest spatio-temporal variation in temperature and wind speed.

8:50

3aPA2. Acoustic travel-time tomography of the atmospheric surface layer at the Boulder Atmospheric Observatory. Vladimir E. Ostashev (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@noaa.gov), Sergey N. Vecherin, D. K. Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), and Alfred J. Bedard (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Boulder, CO)

Acoustic tomography of the atmospheric surface layer (ASL) is based on travel-time measurements between speakers and microphones in a spatial array, which are arranged so as to create propagation paths through the region to be sampled. Then, the temperature and wind velocity fields inside the tomographic region are reconstructed by inverting the travel times. Tomography has certain advantages over conventional point measurements, such as spatial averaging and a quadratic growth of the observations relative to the number of sensors. An array built at the Boulder Atmospheric Observatory (BAO) enables horizontal-slice tomography of the ASL at a height of

8 m above the ground, in an 80 m x 80 m region. The instrumentation and principle of operation of the BAO tomography array are explained. Inverse algorithms for reconstruction of the temperature and wind velocity fields from the travel times are reviewed. Results in numerical simulations of the BAO tomography array and reconstruction of turbulence fields in tomography experiments are presented. Acoustic tomography of the atmosphere can also be performed at other spatial scales, ranging from a size of an ultrasonic anemometer/thermometer to the height of the atmospheric boundary layer and even in the stratosphere and thermosphere.

9:10

3aPA3. Pulse scattering in a turbulent atmosphere with applications to acoustic remote sensing of the atmosphere with sodars. Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@noaa.gov), Stuart Bradley (Phys. Dept., Univ. of Auckland, Auckland, New Zealand), and D. K. Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

A new, rigorous theory of acoustic pulse scattering in a turbulent atmosphere with spatial-temporal fluctuations in temperature and wind velocity is developed. The theory generalizes the classical theory of sound scattering to broadband signals and coupled spatial-temporal fluctuations of random fields. The scattered sound field is obtained as a Born approximation of a set of equations for the sound pressure and acoustic particle velocity. The spatial-temporal correlation function of the scattered field is calculated. These results are obtained without using the quasi-static approximation, which is employed in the classical theory, and enable analysis of this approximation. The spatial, temporal, and frequency coherences of the scattered signal are studied. The results obtained are applied to acoustic remote sensing of the atmosphere with sodars. It is proposed that the frequency spectrum of the temporal correlation function could be used to measure the wind velocity and the variance of the convective velocity fluctuations. An *ad hoc* approach for increasing the spatial resolution in acoustic sounding of the atmosphere is suggested. The theory developed is rather general and applicable to acoustic pulse scattering in other media such as the ocean with temperature, salinity, and current velocity fluctuations.

9:30

3aPA4. Automated detection and cataloging of global explosive volcanism using the International Monitoring System infrasound network. Robin S. Matoza (Dept. of Earth Sci., Univ. of California, Santa Barbara, Webb Hall MC 9630, Santa Barbara, CA 93106, matoza@geol.ucsb.edu), David N. Green (AWE Blacknest, Reading, United Kingdom), Alexis Le Pichon (CEA/DAM/DIF, Arpajon, France), David Fee (Wilson Alaska Tech. Ctr. and Alaska Volcano Observatory, Univ. of Alaska Fairbanks, Fairbanks, AK), Peter Shearer (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Pierrick Mialle (CTBTO, Vienna, Austria), and Lars Ceranna (BGR, Hannover, Germany)

Explosive volcanic eruptions are among the most powerful sources of infrasound observed on earth, with recordings routinely made at ranges of hundreds to thousands of kilometers. These eruptions can also inject large volumes of ash into heavily traveled aviation corridors, thus posing a significant societal and economic hazard. Detecting and counting the global occurrence of explosive volcanism helps with progress toward several goals in earth sciences and has direct applications in volcanic hazard mitigation. This project aims to build a quantitative catalog of global explosive volcanic activity using the International Monitoring System (IMS) infrasound network. We are developing methodologies to search systematically through IMS infrasound array detection bulletins to identify signals of volcanic origin. We combine infrasound signal association and source location using a brute-force, grid-search, cross-bearings approach. The algorithm corrects for a background prior rate of coherent infrasound signals in a global grid. When volcanic signals are identified, we extract metrics such as location, origin time, acoustic intensity, signal duration, and frequency content, compiling the results into a catalog. This work represents a step toward the goal of integrating IMS data products into global volcanic eruption early warning and notification systems.

9:50–10:10 Break

10:10

3aPA5. An innovative technique to probe the middle and upper atmosphere. Jelle D. Assink, Pieter Smets, Läslo Evers (R&D Seismology and Acoust., KNMI, PO Box 201, De Bilt 3730 AE, Netherlands, jelle.assink@knmi.nl), and Alexis Le Pichon (CEA, DAM, DIF, Arpajon, France)

The middle atmosphere has gained more and more importance for the purpose of weather and climate prediction, since increasing evidence indicates that the troposphere and stratosphere are more closely coupled than assumed before. Significant effort has been made toward a more comprehensive representation of the atmosphere to better capture the stratospheric variability as well as the stratospheric-tropospheric interactions, for example, during Sudden Stratospheric Warming (SSW) events. Despite these advances, the upper layers of the atmosphere have remained a region that is difficult to monitor. Over recent years, new developments in the field of infrasound have lead to an innovative method for evaluating numerical weather prediction models. In this presentation, the general technique will be described and a case study will be presented in which stratospheric forecasts of the 2013 major SSW are evaluated.

10:30

3aPA6. Comparison of surface wind noise predictions from mirror flow and rapid-distortion models of atmospheric turbulence. Gregory W. Lyons (National Ctr. for Physical Acoust., The Univ. of MS, 145 Hill Dr., P.O. Box 1848, University, MS 38677-1848, gwlyons@go.olemiss.edu), Jiao Yu (Dept. of Phys., Liaoning Shihua Univ., Fushun, Liaoning, China), and Richard Raspet (National Ctr. for Physical Acoust., The Univ. of MS, University, MS)

For atmospheric acoustic measurements at the ground surface, the principal source of the intrinsic wind noise is shearing of the turbulence by the mean flow. The pressure spectra from this turbulence-shear mechanism depend on two-point statistics of the anisotropic, inhomogeneous atmospheric turbulence. The inhomogeneity of the surface-normal velocity component can be realistically modeled by the mirror flow, which is a superposition of two correlated isotropic turbulent fields in transformed coordinates. Another analytical framework is rapid-distortion theory, which approximates the turbulence as the result of linearized distortion of an initially

homogeneous field by the mean flow. This study compares turbulence-shear spectra calculated with the mirror flow model and rapid-distortion models for both surface blocking effects and mean shear distortion. For each case, the model parameters are estimated by fits to the single-point velocity spectra recorded in a recent experiment near Laramie, Wyoming. The subsequent model predictions for the wind noise spectra are compared with simultaneous measurements from flush-mounted infrasound sensors. [Work supported by the U.S. Army Research Laboratory (Grant No. W911NF-13-2-0021) and the National Natural Science Foundation of China (Grant No. 11304137).]

10:50

3aPA7. Results from the Humming Roadrunner ground truth experiment. Roger M. Waxler, Carrick Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu), David Green (AWE Blacknest, Reading, United Kingdom), Jean-Marie LaLande (NCPA, Univ. of MS, University, MS), and Doru Velea (Leidos, Inc., Reston, VA)

In August 2012, a ground truth infrasound experiment was performed in the American Southwest. Six large chemical explosions were detonated at the White Sands Missile Range in New Mexico. In TNT equivalents, there were two 10 ton charges, two 20 ton charges, and two 50 ton charges. In addition to an extensive near-field deployment to capture source waveforms, a large network of infrasound sensor arrays was deployed at ranges from about 50 km to about 500 km and along a variety of azimuths. Results from the experiment will be discussed.

Contributed Paper

11:10

3aPA8. Development of an anechoic wind tunnel. Salvador Mayoral and Syed Zeeshan Khader (Mech. Eng., California State Univ., Fullerton, 800 N State College Blvd., Fullerton, CA 92831, smayoral@fullerton.edu)

The development of an anechoic wind tunnel is presented. The test section walls of a low-speed open-circuit wind tunnel have been acoustically treated in order to simulate an acoustically free field environment with forward flight. The dimensions of the test section are 0.86 m wide by 0.56 m tall and 1.32 m long. The four walls of the test section are lined with pretensioned Kevlar panels that permit the transmission of sound with less than 2 dB of attenuation. The top and bottom walls of the test section sandwich 4-in. thick acoustic foam between the pretensioned Kevlar and a wooden frame. The side walls of the test section consists of pretensioned Kevlar screens that separate the test section flow from adjacent anechoic chambers. In this fashion, the acoustic measuring equipment is not subjected to the flow inside the test section. Noise sources are modeled with a small loudspeaker that is connected to a signal generator and acoustic measurements are taken with quarter-inch condenser microphones. Preliminary results indicate that the test section is anechoic for frequencies greater than 1 kHz. This work ultimately aims to experimentally investigate the interaction between aeroacoustic noise sources and a momentum wake.

Session 3aPP

Psychological and Physiological Acoustics: Quantitative Methodology in Both Physiological and Psychophysical Data Analysis Workshop

Daniel McCloy, Cochair
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Ross K. Maddox, Cochair
Univ. of Washington

Hari M. Bharadwaj, Cochair
Athinoula A. Martinos Center for Biomedical Imaging, Massachusetts General Hospital, 149 Thirteenth Street, Boston, MA 02129

Chair's Introduction—7:55

Invited Papers

8:00

3aPP1. Using detection theory to analyze human decision making and sensory processing. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

On the 50th anniversary of the publication of Green and Swets' book "Signal Detection Theory and Psychophysics," it is fitting to revisit the question of how detection theory models of human decision making can improve our understanding of sensory processing. Classical psychophysics, as conceived by Fechner in 1850, trusts the observer to report sensory experiences without bias. The innovation of signal detection theory (SDT) was to develop methods that do not require the observer to have introspective awareness of internal sensations. Since its introduction, SDT has inspired experimentalists in many fields, and so now it is appropriate to simply refer to "detection theory" (DT) as the models are appropriate to a wide range of experiments, many of which do not involve signals in noise per se. Data from a previously published intensity discrimination task relying upon working memory (Gallun *et al.*, 2012) will be examined in greater detail to show the techniques and benefits of a detection theory approach to the analysis of human decision making.

8:30

3aPP2. Monte-Carlo analysis of two logical premises to avoid the Probit algorithm for determination of sensory threshold by psychophysics. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

An often confusing and usually avoided, Probit algorithm, is traditionally needed to iteratively calculate an optimal regression line for stimulus strength versus detection rate for psychophysical data that actually appears non-monotonous but supposed to be monotonous. Two logical premises are presented that enable monotonicity even with non-monotonous observed rates: (1) If a trial stimulus is above the threshold then a set of higher stimuli are detectable. (2) If a trial stimulus is below the threshold then a set of lower stimuli are undetectable. The time series in Fig1 has 15 trials of a conventional audiometric threshold tracking session, with 10 dB down for detecting and 5 dB up for missing a stimulus. Following from the two premises, the set of additional points are the predicted responses. Monte-Carlo simulation demonstrated that this strategy not only finds a monotonic relation between detection rate versus stimulus intensity, but also significantly improves accuracy of estimation of the 50% threshold level. In conclusion, two logical premises can help to obtain uniform monotonicity of the sigmoid shaped psychometric regression curve, even without using conventional probit iterations, leading to faster and simplified estimation of the 50% threshold point with greater accuracy. Further research is needed to evaluate clinical impact of this concept for improving current procedures of threshold estimation without increasing the number of actual observations for various psychophysical assessment of sensory thresholds.

9:00

3aPP3. Unified analysis of accuracy and reaction times via models of decision making. Samuel R. Mathias (Dept. of Psychiatry, School of Medicine, Yale Univ., 2 Church St. South, New Haven, CT 06519, samuel.mathias@yale.edu)

"Sequential-sampling" models aim to characterize decision making during forced-choice experiments. These models decompose the decision process into its constituent elements, and can explain not only the probabilities with which subjects make each possible response, but the full distributions of reaction times associated with each of the response alternatives. Thus, such models provide a powerful framework for the analysis of behavioral data. However, despite being extremely popular in the cognitive sciences, they are almost never used in psychoacoustics. By way of example experiments, I will discuss how to fit these kinds of models to psychoacoustical data, using traditional maximum likelihood and hierarchical Bayesian inference, and how to interpret their results in terms of the underlying decision processes.

9:30

3aPP4. Generalized linear mixed models in hearing science. Hari M. Bharadwaj (Athinoula A. Martinos Ctr. for Biomedical Imaging, Massachusetts General Hospital, 149 Thirteenth St., Boston, MA 02129, hari@nmr.mgh.harvard.edu)

Linear mixed-effects models (LMEs), by virtue of allowing for the quantification of both random effects (e.g., associated with sampling of subjects) and the effects of fixed predictor variables, provide a powerful statistical modeling framework that can accommodate a large number of common experimental designs used in hearing science. Examples of designs for which LMEs are suitable include repeated-measures designs, longitudinal studies, and multilevel designs. An extensive literature of robust methods exists for fitting such models for normally distributed responses. On the other hand, often, data in hearing science experiments are binary (e.g., detected or missed), proportions, or counts (number of correct responses) and are better modeled using non-normal distributions (e.g., binomial). Generalized linear mixed models (GLMMs) provide a framework to retain the advantages of LMEs in modeling random effects by modeling the non-normal response variables using LMEs through (nonlinear) link functions. In this presentation, using example simulated and real psychophysical (attention task performance) and physiological data (OAEs, EFRs), I will introduce GLMMs and discuss the great flexibility they offer, and the challenges involved in parameter estimation and inference. I will also point to emerging methodological options and software packages for using GLMMs.

10:00–10:15 Break

10:15

3aPP5. Advances in quantifying listening effort: Growth curve analyses of pupillometry data. Stefanie E. Kuchinsky (Ctr. for Adv. Study of Lang., Maryland Neuroimaging Ctr., Univ. of Maryland, Maryland Neuroimaging Ctr., 8077 Greenmead Dr., Bldg. #795, College Park, MD 20740, skuchins@umd.edu), Judy R. Dubno, and Mark A. Eckert (Dept. of Otolaryngol. - Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Understanding speech in background noise often requires substantial mental effort, especially in adverse signal-to-noise ratios (SNRs) and for older adults with hearing loss. Pupillometry has been used to index listening-related effort, because pupil dilation is an autonomic response that is modulated by changes in cognitive demands, such as when recognizing speech in decreasing SNRs. This presentation will focus on our approach to quantifying the pupil response through growth curve analysis (GCA), a multilevel modeling technique in which orthogonalized polynomials are fit to time series data to capture its shape. We will provide an overview of our pupillometry research that employed GCA to characterize (1) changes in the pupil response across varying listening conditions and (2) individual differences in the pupil response among older adults with hearing loss. Discussion will include statistical advantages of this approach compared to commonly used peak-picking analyses, implementation of GCA in R, and the theoretical interpretation of model results. We will conclude by demonstrating that GCA results can be linked to functional neuroimaging data in order to further our understanding of the neural mechanisms that underlie listening effort. [Work supported, in part, by NIH/NIDCD and a Hearing Health Foundation Centurion Clinical Research Award.]

10:45

3aPP6. Individual differences in hearing-impaired data: Stats, troubles, and approaches. Sarah Verhulst (Medizinische Physik and Cluster of Excellence Hearing4all, Oldenburg Univ., 677 Beacon St., Boston, MA 02215, save@bu.edu)

Individual differences in hearing ability might be dominated by subcomponents of hearing loss, e.g., cochlear gain loss, cochlear neuropathy, temporal coding deficits in low/high frequency regions, or combinations of these components. Unfortunately, we can only rely on indirect and hypothesis-driven objective (e.g., OAE/ABR/EFR) and psychoacoustic threshold metrics that aim to quantify these subcomponents of hearing loss, complicating a straightforward explanation of study results. Because correlations statistics often rely on small listener groups in which each data point could have resulted from different SNRs, metric-specific variability, it is not always clear which correlations are significant and meaningful. Additionally, multiple measures provide a multitude of correlations that should all support the common underlying hypothesis before conclusions can be drawn. In this tutorial, I provide some examples and approaches to more *(and less)* meaningful correlations based on recently collected objective and psychoacoustic measures in a group of normal and hearing-impaired listeners. Finally, I will introduce how computational model approaches might direct the interpretation of experimental results when several interacting sources of hearing impairment impact outcome measures unexpectedly.

3a WED. AM

11:15

3aPP7. Nonparametric statistical approaches to neuroimaging data. Abigail L. Noyce (Boston Univ., 2 Cummington Mall, Boston, MA 02215, anoyce@bu.edu) and Robert Sekuler (Brandeis Univ., Waltham, MA)

Neuroimaging, including electroencephalography (EEG), magnetoencephalography (MEG), and functional magnetic resonance imaging (fMRI), is a rich source of information, allowing perceptual researchers measure neural responses to sensory inputs. Scalp EEG's high temporal resolution makes it a method of particular interest to auditory scientists who want to identify (and possibly model) the spatial and temporal extent of a difference between the two conditions. Analyzing neuroimaging data, however, poses several challenges. First, collecting data from tens or even hundreds of sensors and hundreds of time points creates a multiple comparisons problem. Second, neuroimaging data are often quite noisy (as is the brain itself). Finally, imaging data distributions are often incompatible with the assumptions of traditional parametric statistics—most notably, data are often non-normal, and with unequal variance. We illustrate strategies for addressing these challenges in the context of a selective attention study. EEG was recorded while subjects attempted to report a central target in the presence of either standard or oddball distractors. Using clustering and Monte Carlo permutation, we showed that neural responses to the distractors elicit an oddball effect.

WEDNESDAY MORNING, 25 MAY 2016 SNOWBIRD/BRIGHTON, 8:30 A.M. TO 11:55 A.M.

Session 3aSA

Structural Acoustics and Vibration: Computational Methods in Structural Acoustics and Vibration

Robert M. Koch, Chair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Chair's Introduction—8:30

Invited Papers

8:35

3aSA1. Design optimization of a stiffened flexible panel under turbulent boundary layer excitation. Kuangcheng Wu, Nicholas Stowe, and Eric Brown (Ship Survivability, Newport News ShipBldg., 4101 Washington Ave., Newport News, VA 23693, kcwu@msn.com)

The vibration and noise generated from turbulent boundary layer (TBL) flow are main design considerations in the Automotive, Aerospace, Shipbuilding industries, as well as many others. Simulation has been widely applied to help reduce TBL-induced vibration and noise; however, a fully coupled CFD + FEA analysis is still a computational challenge for high-fidelity models. On the other hand, when the block pressure assumption is applicable, a general approach applies the fluctuating pressure underneath the TBL as excitation to the structure to predict its vibratory response and noise. This paper first uses the general approach to predict the vibration and noise of a stiffened flat panel subjected to TBL excitation. The various methods, both empirical and numerical, that are used to estimate the TBL wall spectrum are outlined. Structural transfer functions from Energy Finite Element Analysis (N. Vlahopoulos, K. Wu, MAST Americas 2010) are calculated for the panel. The calculated TBL excitation (M. Goody, AIAA, 2004) and structural transfer functions are combined to predict the flow-induced vibration and noise. In addition, optimization algorithms are applied to evaluate the impacts of several parameters (i.e., varied plating thickness and local damping) and determine the best design of the panel within the set of parameters.

8:55

3aSA2. Direct and transient analyses of the Schur complement for complex vibrating systems. James G. McDaniel, Andrew S. Wixom, and Rodolfo Rodriguez (Dept. of Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jgm@bu.edu)

We are interested in computing the steady-state response of coupled vibrating systems over a frequency band. For each system, the present coupling analysis requires the impedance matrix that relates forces and velocities at the coupled or externally forced degrees of freedom at all frequencies of interest. Algebraic manipulation of each system's full impedance matrix yields the desired impedance matrix in the form of a Schur complement. The present work surveys and compares two approaches for computing the Schur complement, with a particular focus on the effects of damping. The direct analysis computes the Schur complement in the frequency domain, which requires a linear solve of a large system with several forcing vectors at each frequency. The transient analysis computes the impulse responses of the coupled degrees of freedom by numerical integration. This is followed by a Fourier transform that yields the Schur complement. Approaches for choosing the most efficient analysis are presented, along with numerical examples that illustrate and compare both approaches.

9:15

3aSA3. Anticlastic bending of rectangular plates and inference of Poisson's ratio. Micah R. Shepherd and Stephen A. Hambric (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

Elastic plates will exhibit anticlastic behavior due to the Poisson effect. This effect is often overlooked in the analysis of rectangular plates. However, the effect can be substantial for when the boundary conditions are free, causing nodal lines of certain mode shapes to be curved rather than straight. A rectangular plate finite element model comprised of quadratic solid elements was exercised to determine the dependency of the anticlastic bending on plate thickness, aspect ratio, and elastic modulus. A metric to quantify anticlastic behavior was then developed for both bending modes and axial modes. Finally, experimentally obtained mode shapes were used to infer the Poisson's ratio of a rectangular section of hard foam using the derived dependencies.

9:30

3aSA4. Prediction of broadband high-frequency acoustic reflection and transmission from elastic plates with structural discontinuities. Mauricio Villa and Donald Bliss (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Box 90300 Hudson Hall, Durham, NC 27708, mauricio.villa@duke.edu)

A model for the acoustic reflection and transmission for flexible boundaries with structural discontinuities is developed. The system studied is an infinite, fluid loaded plate with periodically fixed constraints that is driven by an incident acoustic field. To treat the local discontinuities of the plate, the coupled structural/acoustic problem is treated by applying the method of Analytical-Numerical Matching (ANM). The ANM framework, which separates the problem into a global numerical solution and a local analytical solution, offers an approach to handle mathematical difficulties associated with the coincidence frequencies of the system. Accurately treating the rapid spatial variation around the discontinuities improves the overall accuracy and convergence of the computational analysis to predict the resulting pressure levels. Results for an infinite plate with discontinuities are compared to a baffled plate with a fluid loading correction introduced to the structural wavenumber. Simple closed-form band-averaged directivity patterns are determined. This approach facilitates an energy-intensity reformulation of the structural and acoustic equations, allowing for the possibility of formalizing the couplingbetween energy flows in the acoustic and structural systems. The overall goal is to develop efficient first principles computational models for more accurate modeling of broadband structural-acoustic reflection and transmission between coupled acoustic spaces.

9:45

3aSA5. Partial differential equation-constrained optimization framework for inverse design of acoustic damping layers. Clay Sanders, Wilkins Aquino (Dept. of Civil and Environ. Eng., Duke Univ., Durham, NC, clay.sanders@duke.edu), and Timothy F. Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., Albuquerque, NM)

We present a partial differential equation-constrained framework for the design of viscoelastic damping layer assemblies in one and two dimensions. Viscoelastic foams' responses to loads not only provide stiffness and damping, but also vary with frequency. In a finite-element formulation, a viscoelastic solid's structural stiffness is determined by its complex valued shear and bulk moduli. By optimally selecting the moduli values of a layered assembly, we demonstrate an effective design of graded foam that minimizes its acoustic scattering in a fluid waveguide. We define a cost functional based on the scattered field from the acoustic-structural interaction of the viscoelastic inclusion in a fluid medium. In the optimization calculations, the complex valued moduli were modified to minimize the objective functional representing the scattered field energy. Models used in numerical simulations featured circular inclusions with distinct concentric rings. This talk will focus on the convergence of optimal designs and effects of frequency and solver algorithms. Numerical results suggest the potential for optimally designed viscoelastic foams to minimize acoustic scattering. [Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL850000.]

10:00

3aSA6. Modeling and design of lightweight, hyperdamping metamaterials for large, broadband vibroacoustic energy attenuation. Ryan L. Harne, Yu Song, and Quanqi Dai (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., E540 Scott Lab, Columbus, OH 43210, harne.3@osu.edu)

The absorption and dissipation of spectrally broadband vibration and wave energy are long-standing pursuits for researchers and engineers. In vehicular applications, light-weighting demands have led to the common use of flexible structural components that exacerbate concerns of low frequency energy transmission and radiation while joints and geometrical features result in higher frequency vibrations that produce adverse noise fields for occupants. To address the needs for broadband vibroacoustic energy attenuation via a lightweight material solution, this research integrates concepts from recent studies on elastic and poroelastic composite materials and investigates a new idea to cultivate hyperdamping effects in an engineered metamaterial. A finite element (FE) model is developed to identify ways in which to effect hyperdamping properties by virtue of controlled instability mechanisms embedded within the metamaterial. Throughout the design space, the simulations predict numerous design parameter selections that generate hyperdamping properties for large, broadband energy trapping and dissipation. Experimental studies verify the trends of FE model predictions and exemplify the potentials for hyperdamping phenomena once leveraged in practice.

10:15–10:25 Break

10:25

3aSA7. When an offshore cylindrical pile is impacted axially, the resulting wall vibration and underwater radiated sound pressure are insensitive to which thin shell theory is selected. Marshall V. Hall ((retired), 9 Moya Crescent, Kingsgrove, NSW 2208, Australia, marshallhall@optushome.com.au)

When a pile is impacted axially, a pulse of axial and radial vibration travels downwards and undergoes continued reflections at the toe and head. If a pile is to be treated as a thin cylindrical shell then producing a model for radiated sound pressure requires a particular "thin shell theory" to be selected. There are 12 such theories, as catalogued in Leissa's monograph "Vibration of shells." Four have been selected: the Membrane, Donnell-Mushtari, Flugge, and Epstein-Kennard theories (the last is the most intricate of the 12). An offshore pile with 25.4-mm wall thickness and 381-mm radius (6.7%-ratio) is modeled. Since the pile head is above the water surface, aerial radiation strongly attenuates the vibration of the immersed pile near the pile's ring frequency (2.1 kHz). Spectra of radial vibration and underwater radiated sound pressure are computed up to 10 kHz using each of the four theories and found to be the same, except near the ring frequency. When the calculations are repeated for a similar pile with a wall three times thicker (no longer a thin shell), the Flugge results are higher than the other three by an amount that increases to 2 dB as frequency increases to 10 kHz.

10:40

3aSA8. Effects of internal acoustic coupling on the response of a base-excited hollow structure. Ryan Schultz and Greg Tipton (Analytical Structural Dynam., Sandia National Labs., 1701 Singletary Dr. NE, Albuquerque, NM 87112, rschult@sandia.gov)

Traditionally, structural dynamic simulation predictions for hollow structures do not include internal acoustic volumes. This is the case for two reasons. First, it can be difficult to create a volume mesh of complicated

3a WED. AM

internal cavities, and this additional volume would drastically increase the size of the model. Second, it is often blindly assumed that acoustic coupling is not important for structures with anything but thin walls. However, if the structural and acoustic modes are compatible, that is, the modes have similar frequencies and shapes, then the interaction between the structure and the enclosed acoustic volume will occur and can have an effect on structural response. The magnitude of this coupling effect is demonstrated here by examining the structural response predictions for a finite element model of a hollow structure with and without the acoustic volume in place. When a base excitation is applied at a frequency and location to excite compatible modes, the response predictions are seen to be drastically different if the acoustic volume is included. The frequency response results of the coupled finite element simulations will be compared with those of the classic two degree of freedom dynamic vibration absorber, showing the effect of frequency ratio and damping on the structural response.

10:55

3aSA9. Energy flux streamlines for an ensonified fluid sphere near a rigid boundary: Implementation and testing. Cleon E. Dean (Phys., Georgia Southern Univ., PO Box 8031, Math/Phys. Bldg., Statesboro, GA 30461-8031, cdean@georgiasouthern.edu) and James P. Braselton (Mathematical Sci., Georgia Southern Univ., Statesboro, GA)

Progress in the implementation and testing of routines to compute the energy flux [J. A. Mann III, *et al*., J. Acoust. Soc. Am. **82**, 17–30 (1987)] and energy flux streamlines [D. M. F. Chapman, J. Acoust. Soc. **124**, 48–56 (2008)] for the problem of an ensonified fluid sphere near a rigid boundary are presented. The problem is isomorphic via the method of images to the problem of interactive scattering between two identical fluid spheres. Trinks provided an analytical solution to this problem in 1935 [W. Trinks, Ann. Phys. **414**, 561–590 (1935)]. However, his solution is mathematically intractable for larger spheres due to the complexity of his translational addition theorem for spherical harmonics. Later, researchers Cruzan [Quart. Appl. Math. **20**, 33–40 (1962)] and Liang and Lo [Radio Science 2, 1481 (1967)] simplified this translational addition theorem, making it feasible to solve the problem with larger spheres. This work follows Bruning and Lo [Tech. Rep. 69-5 (Antenna Laboratory, Univ. of Illinois, Urbana, IL 1969)], correcting typos in that work to implement efficient algorithms for acoustical scattering.

11:10

3aSA10. Numerical prediction and analysis of electromagnetic vibration and noise of claw pole alternator. Shuanglong Wu, Shuguang Zuo, and Xudong Wu (Clean Energy Automotive Eng. Ctr., Tongji Univ., No.4800 Cao'an Rd., Jiading District, Shanghai 201804, China, zymwgl@foxmail.com)

With the improvement of internal combustion engine noise, the noise of claw pole alternator which is universally used on modern automobiles is becoming more distinct. Extensive experimental results show that the noise of claw pole alternator is annoying at low to middle speed and mainly originates from the electromagnetic noise. In this paper, a complete methodology for electromagnetic vibration and noise prediction of claw pole alternator is presented. It is a three-step multiphysics simulation. The first step is to calculate the magnetic force exerted on the inner surface of the stator using 3D FEM. The second step is the structural analysis, which is to investigate the natural resonant frequencies and modal shapes of the alternator system taking into account the real mechanical boundary conditions. The third is the electromagnetic vibration and noise calculation by using modal superposition method and boundary-element method, respectively. Simulation results agree well with those measured by experiments. Furthermore, the mechanisms of electromagnetic vibration and noise are analyzed by using the electromagnetic theory. The method can be used in the initial design stage of claw pole alternator and is also suitable for other types of motor.

11:25

3aSA11. Calculation of sound radiation in infinite domain using a meshless method. Shaowei Wu and Yang Xiang (School of Energy and Power Eng., Wuhan Univ. of Technol., Peace Ave., Wuhan, Hubei Province, No. 1040, Wuhan, Hubei 430063, China, thinkwsw@qq.com)

A meshless method coupling with a variable order infinite acoustic wave envelope element for sound radiation calculation in infinite domain is presented with the aim of accurately calculating the acoustic radiation and improving computational efficiency. It is based on using the element-free Galerkin method in the inner region enclosing the radiator and a variable order infinite acoustic wave envelope element in the outer region for the proper modeling of the pressure amplitude decay. The details are provided for the derivation and implementation of this method. The factors of influencing the performance of the method, which include the shape function constructing, the number of integration points, the weight functions, and the support domain, are discussed. A hybrid adaptive Gauss-Legendre quadrature is devised to obtain good integration accuracy. The suitable radius of the support domain for the acoustic field calculation in free space is also determined by use of numerical experiments. A complex structure is designed for simulation to validate the method. The results illustrate the accuracy, applicability, and effectiveness of this method.

11:40

3aSA12. Analytical prediction of radiation from a multi-element variable-directivity spherical loudspeaker array and construction and measurement of a prototype. Taylor R. Laub and Donald B. Bliss (Mech. Eng., Duke Univ., 148B Hudson Hall, Duke University, Durham, NC 27705, dbb@duke.edu)

An analysis of radiation from a spherical surface with twelve radiating spherical caps oriented in a dodecahedral arrangement is performed. The approach generalizes the classical single cap solution by utilizing spherical trigonometry transformations to create many spherical caps in different orientations. The calculations include far-field directivity, surface impedances, and net radiated power for a variety of cap motions within the spherical array. Simulations show how well the discrete caps can produce monopole, dipole, and higher mode directivity patterns. Relative motions of the caps are calculated that best simulate these individual radiation modes. The frequency ranges for accurate reproduction of these radiation modes using discrete caps are quantified, and the role of large cap sizes to extend the range is shown. At higher frequencies, the phase interference between discrete cap radiators cause the intended radiation patterns to break down in complicated ways. A prototype 12-element spherical speaker enclosure has been constructed, with the help of a 3-D printer, and tested in an anechoic chamber. The configuration of typical cone loudspeakers creates constraints on the practical design of spherical speakers, limiting the radiating area fraction. Driver design changes to allow more compact spherical speakers with extended frequency range are suggested.

Session 3aSC

Speech Communication and Psychological and Physiological Acoustics: Gender Effects in Speech Production and Perception

Sarah H. Ferguson, Cochair
Communication Sciences and Disorders, University of Utah, 390 South 1530 East, Room 1201, Salt Lake City, UT 84112

Eric J. Hunter, Cochair
Department of Communicative Sci., Michigan State University, 1026 Red Cedar Road, East Lansing, MI 48824

Chair's Introduction—8:00

Invited Papers

8:05

3aSC1. Gender differences and speech accommodation in occupational settings. Eric J. Hunter (Dept. of Communicative Sci., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ejhunter@msu.edu), Sarah Hargus Ferguson (Dept. of Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT), Tim Leishman (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lynn Maxfield (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT), Simone Graetzer, and Pasquale Bottalico (Dept. of Communicative Sci., Michigan State Univ., East Lansing, MI)

Nearly one quarter of the U.S. workforce depends on a healthy, versatile voice as a tool for their profession. These are individuals who, lose voice quality and/or vocal endurance, would not be able to perform their job effectively. These *occupational voice users* include professionals such as teachers, counselors, emergency dispatchers, air traffic controllers, performers, and telephone workers. Women tend to have a disproportionate incidence of reported voice problems compared to men. They also make up the majority of several of these high voice-use occupations (e.g., public school teachers, call center workers). This presentation will provide an overview of our current understanding of gender discrepancy in vocal health issues as well as a discussion of recent results identifying underlying causes, which may contribute to their heightened risk. Such results include compensatory adjustments women use in different communication environments, speech accommodation to stress, and the relationship between vocal fatigue and pulmonary function.

8:25

3aSC2. The effect of compromised pulmonary function on speech production among female school teachers. Lynn M. Maxfield (National Ctr. for Voice and Speech, Univ. of Utah, 136 S Main St., Ste #320, Salt Lake City, UT 84101, lynn.maxfield@utah.edu), Eric Hunter, and Simone Greatzer (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Females face a significantly higher risk than males of developing long-term voice problems with lifetime instances occurring in 46% of females compared to 37% of males. The higher incidence of prolonged problems among women has been associated with a number of gender differences, including physiological differences in the laryngeal system, differences in the endocrine system, and differences in pulmonary usage. Additionally, inefficient pulmonary utilization and reduced lung volume have been linked with vocal health concerns. Our study sought to use established spirometry measures and a relatively new questionnaire, the Vocal Fatigue Index (VFI), to determine if there is a relationship between pulmonary function and vocal fatigue among teachers. Additionally, if there is a relationship, to determine if that relationship is stronger in females than in males. **122** (96 females, 26 males) elementary and middle school teachers from the Jordan School District in northern Utah participated in this research. For females, VFI was a predictor of several spirometry measures; however, the same correlation was not found among male participants. These results indicate that reduced pulmonary function in combination with other gender differences in speech production may lead to increased incidences of vocal fatigue among female teachers than their male counterparts.

8:45

3aSC3. Phonetic convergence and talker sex: It's complicated. Jennifer Pardo, Adelya Urmanche, Sherilyn Wilman, and Jaclyn Wiener (Psych., Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, pardo@optonline.net)

Investigations of phonetic convergence report conflicting results with respect to talker sex. Some studies report that females converge to a greater degree than males, while others find no difference or the opposite pattern. These discrepancies frustrate attempts to characterize the impact of talker sex on phonetic variation and convergence in a straightforward manor. The current investigation reveals that talker sex interacts with other variables, both lexical and phonological. A set of 92 talkers (47 females) shadowed monosyllabic words that manipulated word frequency within eight vowels. Phonetic convergence was assessed in an AXB perceptual similarity task and in F1 x F2 vowel space. Convergence in F1 x F2 vowel space did not differ between males and females on average, but female

talkers converged to front vowels (/i/, /ɛi/, /ɛ/, /æ/) more than to back vowels (/ɑ/, /ou/, /ʊ/, /u/), and male talkers showed the opposite pattern. Furthermore, higher vowels (/i/, /ɛi/, /ʊ/, and /u/) showed the largest differences in convergence between men and women. These patterns were largely driven by convergence on F2 alone. These findings relate to broader sociolinguistic concerns about the impact of gender on phonetic variation and sound change.

9:05

3aSC4. Same versus opposite-sex accommodation in digital media speech. Tanya Flores (Lang. and Lit., Univ. of Utah, 255 S Central Campus Dr., LNCO 1400, Salt Lake City, UT 84109, Tanya.Flores@utah.edu)

This presentation focuses on the linguistic and social factors that motivate phonetic variation of the /tɾ/ cluster in Chilean Spanish, specifically producing the cluster as an alveo-palatal affricate [tʃ] (or with rhotic [tʃɾ]). The data consist of 1,596 tokens produced by 72 Santiago speakers in a corpus of digital radio recordings. Results from a mixed-effects logistic regression analysis confirm that several linguistic and social factors, including speakers' sex and age, affect /tɾ/ cluster variation in this dataset. Using a speech accommodation analysis, the study allows for an examination of the effect that social traits of the direct addressee (co-host or in-studio guest) have on speakers' variant choice and usage rates. Results from the two-population binomial tests on speaker-listener pairings reveal that the affricated variant [tʃ] is significantly affected by the age of the addressee. Moreover, opposite-sex pairs had a significantly higher [tʃ] production rate than same-sex pairs or solo hosts of either sex. Specific patterns by sex will be elaborated in this presentation.

9:25

3aSC5. Quantity of mothers' and fathers' speech to sons and daughters. Mark VanDam (Dept. of Speech and Hearing Sci., Elson S. Floyd College of Medicine, Washington State Univ., PO BOX 1495, Spokane, WA 99202, mark.vandam@wsu.edu) and Tracy Tully (Dept. of Commun. Disord., Eastern Washington Univ., Spokane, WA)

The literature on child-directed speech has shown that mothers, on average, talk to their children with higher token frequency than fathers, but fathers may use more complex language or more variable word types. Other recent research has shown that mothers may use greater fundamental frequency variability and range when talking to their children, as compared with fathers who showed no fundamental frequency differences between child- and adult-directed speech. There is also some evidence that mothers may talk more often with their daughters than sons. Whether fathers are sensitive to the sex of their children is unknown. This work looks at the quantity of fathers' and mothers' speech to their sons and daughters from a large database of daylong recordings collected from a body-worn recorder on preschool boys and girls in their natural daily environments. The recordings were analyzed offline with automatic speech processing software which tallied the quantity of syllables produced for all individual talkers and the conversational exchanges among mothers and fathers with their sons and daughters. Contrary to previous findings, results indicate that fathers may engage more often with their sons than daughters, but mothers are not sensitive to child sex.

9:45

3aSC6. Talker gender effects in the Ferguson Clear Speech Database. Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu)

The Ferguson Clear Speech Database (FCSD; Ferguson, 2004) was designed to support a talker-differences approach to the investigation of the acoustic characteristics that underlie the superior intelligibility of clear speech. However, its size (41 talkers) and balanced gender composition (21 females and 20 males) have made it useful for exploring talker gender effects on a wide array of acoustic and perceptual measures. In this talk, I will present comparisons of male and female talkers from previous studies as well as more recent ones, along with new analyses of old data. I will also consider how the magnitude of any talker gender differences compares to the variability observed within each gender.

10:05–10:20 Break

10:20

3aSC7. Effects of auditory training on cochlear implant users' gender categorization. Bomjun J. Kwon (Hearing, Speech and Lang., Gallaudet University, 800 Florida Ave. NE, Washington, DC 20002, bomjun.kwon@gallaudet.edu) and Qian-Jie Fu (Dept. of Head and Neck Surgery, Univ. of California, Los Angeles, CA)

Nowadays, the benefits of cochlear implants (CIs) for individuals with a substantial degree of hearing loss are widely demonstrated in clinical applications in terms of both speech recognition and speech production. However, a body of evidence indicates CI users' compromised ability to accurately categorize gender of the voice (e.g., Fuller *et al*., J. Assoc. Res. Otol. **15**, 1037–1048, 2014), which may be an obstacle to overcome to improve executive functions for communication with the device. While it has been known that CI users are capable of utilizing the fundamental frequency (F0) of the voice as the primary determinant of gender categorization, poor differential selectivity in F0 with the device attributes to the compromised ability. Another vocal characteristic, vocal tract length (VTL), which is known to play an important role in gender categorization in normal hearing (NH) listeners, is largely ignored in CI users. Considering that the VTL information, grossly reflected on formants, may be transmitted through the device and perceived by CI users when presented in isolation, and that CI users, in general, highly utilize episodic context, proper auditory training of CI users with voice stimuli with F0 and formants covaried or independently varied might improve their gender categorization.

10:40

3aSC8. Perceived gender in clear and conversational speech. Jaime A. Booz and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 2808 S 2700 E, Salt Lake City, UT 84109, jaime.booz@utah.edu)

Much of our understanding of gender perception in voice and speech is based on sustained vowels or single words, which eliminates temporal, prosodic, and articulatory cues available in more natural, connected speech. Further, while many studies have examined acoustic and sociolinguistic differences between male and female voices, the relationship between talker speaking style and perception of gender has not yet been explored. Clear speech, adopted by talkers who perceive some barrier to effective communication, is one such speaking style change. The present study examines the relationship between clear speech and talker gender perception. Clear and conversational neutral sentences produced by all 41 talkers from the Ferguson Clear Speech Database (Ferguson, 2004) were presented to young listeners with normal hearing. They rated the gender of the talker using a visual analog scale with endpoints labeled masculine and feminine, chosen to capture small within-category changes in perceived gender. Acoustic analyses of these sentences, including fundamental frequency, formant frequencies, speaking rate, fundamental frequency range, and cepstral peak prominence will be undertaken to determine the relationship between acoustic correlates of clear speech and listener ratings of femininity or masculinity. Applications to transgender voice therapy will be discussed.

11:00

3aSC9. Jimmy Scott and the problem of gender in singing. Nina Eidsheim and Jody E. Kreiman (UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

The voice of jazz performer Jimmy Scott raises interesting questions of how gender is marked (or not marked) in singing voice. Scott was born with Kallman's Syndrome, which affects male hormonal levels and prevents the onset of puberty. Although he self-identified as a "regular guy," in his career he was presented as a novelty act—a boy who sounded like an adult woman—or paired with gender-ambiguous images. This paper explores Scott's voice in comparison to male and female peers, with emphasis on the paradoxical role of falsetto in creating a male vocal image.

11:20

3aSC10. Are women more unstable than men? Shaky topics. Julie M. Barkmeier-Kraemer (Otolaryngol., Univ. of Utah, 50 N Medical Dr., 3C120, Salt Lake City, UT 84132, JulieB.Kraemer@hsc.utah.edu)

Essential vocal tremor was reported by Sulica and Louis (2010) to predominantly occur in females (90%) compared to males (10%) even though the overall diagnosis of essential tremor demonstrates equal representation of both groups. Interestingly, vocal tremor manifests in 30% of individuals diagnosed with essential tremor as well as spasmodic dysphonia. However, information regarding the proportion of females and males exhibiting vocal tremor in those diagnosed with spasmodic dysphonia remains unclear. A predominance of females represented with isolated vocal tremor and in those diagnosed with spasmodic dysphonia could suggest either a genetic or endocrine system link to the onset of vocal tremor. The purpose of this study was to compare and contrast the representation and characteristics of males and females diagnosed with isolated vocal tremor to spasmodic dysphonia with vocal tremor. Comparisons will be made between speech structures exhibiting tremor on endoscopic examination and vocal tremor acoustic patterns. A preponderance of females with vocal tremor in both groups supports a genetic link to the onset of vocal tremor. Differences in acoustic patterns and structural involvement profiles between males and females would support the possible influence of the endocrine system in the development of vocal tremor.

11:40

3aSC11. The influence of linguistic complexity and gender identity on /s/ variation in children. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, munso005@umn.edu)

Children, like adults, vary in the extent to which their speech conforms to stereotypical expectations for their biological sex. Previous research has examined this in boys with and without gender dysphoria (GD) [Munson *et al*., J. Acoust. Soc. Am. **137**, 1995–2003 (2015); Munson, J. Acoust. Soc. Am. **138**, 1895 (2015)]. Munson (2015) showed that 5–13 year old boys with GD were more likely than age-matched boys without GD to produce tokens of /s/ with a diffuse spectrum, consistent with a frontal, i.e., /θ/-like, /s/. This was based on an analysis of /s/ tokens from single word productions and from fluent sentence repetitions. The purpose of this presentation is to further examine the /s/ differences between these two groups of boys in single words, sentence repetitions, and spoken narratives elicited through a picture-description task. Analyses will examine whether the differences between the boys with and without GD are exaggerated in the narrative data, which offer more freedom for gender performativity than do single words and sentences. We will also examine whether the differences between groups are mediated by individual children's language expressive language abilities, as measured by the objective complexity of their narratives, and by the overall accuracy of sentence repetition.

Session 3aSP

Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics: Acoustic Array Systems and Signal Processing II

John R. Buck, Cochair
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Mingsian R. Bai, Cochair
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Chair's Introduction—8:30

Contributed Papers

8:35

3aSP1. Recording of extended sound fields using spherical microphone arrays and a-priori knowledge of the sound source positions. Jorge A. Trevino Lopez (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 9808577, Japan, jorge@ais.riec.tohoku.ac.jp), Keigo Wakayama (NTT Media Intelligence Labs., NTT Corp., Musashino, Tokyo, Japan), Shuichi Sakamoto, Yo-iti Suzuki (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., Sendai, Miyagi, Japan), Hideaki Takada, and Manabu Okamoto (NTT Media Intelligence Labs., NTT Corp., Musashino, Tokyo, Japan)

The acquisition of comprehensive sound field information is a central topic in spatial acoustics. In conventional systems, recording devices must be located at the listener's viewpoint. This research introduces a sound field recording technique for spherical microphone arrays which makes use of *a-priori* information regarding the distribution of sound sources. The proposal generates sound field descriptions for viewpoints that are located away from the recording device, as long as there are no sound sources between the target viewpoint and the microphone array. Sound field descriptions, a set of spherical harmonic expansion coefficients, are generated from the array signals. A translation operator, calculated from approximate source positions known in advance, is defined so as to shift the plane wave decomposition of the partial fields associated with each expansion coefficient. The proposed method is compared with an existing technique based on a different kind of translation operator traditionally used in implementations of the boundary element method. These operators require no a-priori information, but their region of validity is limited by the spherical harmonic expansion order. The conventional method achieves greater accuracy in the proximity of the recording array; however, the proposed method can generate sound field descriptions covering a much larger region.

8:50

3aSP2. Maximum likelihood estimation to denoising channels in beamforming circular array. Fabricio A. Bozzi, José Manoel de Seixas (Sonar Group, Brazilian Navy Res. Inst., Rua Hugo Leal, Rio de Janeiro 21931250, Brazil, bozzi@ipqm.mar.mil.br), Thiago C. Xavier (LabSonar, Federal Univ. of Rio de Janeiro, Rio de Janeiro, Brazil), and Leonardo M. Barreira (Sonar Group, Brazilian Navy Res. Inst., Rio de Janeiro, Brazil)

The delay and sum beamforming is the most simple technique in direction of arrival (DOA) Estimation. Although its performance on spatial discrimination is poor, compared to other beamforming, delay and sum still is used in large operating sound navigation and ranging (SONAR) because of its low computational cost. A circular hydrophone array (CHA), commonly used in SONAR system, is an attractive alternative to provide a more uniform directive response over all azimuth angle. This array is analyzed here, working with experimental data, acquired in a acoustic tank and in the sea. Maximum likelihood estimation (MLE) is applied to denoising noisy channels, summing up them after in delay-and-sum. First of all, a noise in an acoustic tank is considered to represent the hydrophones, cables, and acquisition system noises. Then, an environmental noise is collected in the sea. The MLE use both of them to calculate the weights in the beamforming. A boat is used to running around the array, and the DOA of the uniformly weight and MLE in delay-and-sum shows the performance improvement.

9:05

3aSP3. Simultaneous tracking and counting of targets in a sensor network. Pritthi Chattopadhyay, Asok Ray (Mech. Eng., The Penn State Univ., University Park, PA), and Thyagaraju Damarla (Army Res. Lab., Networked Sensing and Fusion Branch, U.S. Army Res. Lab., Adelphi, MD 20783, thyagaraju.damarla.civ@mail.mil)

Unattended ground sensors (UGS) are widely used to monitor human activities, such as pedestrian motion and detection of intruders in a secure region. This paper presents an algorithm for counting and tracking humans moving through a UGS network. Each node of this sensor network is equipped with a geophone (i.e., seismic sensor) and a microphone (i.e., acoustic sensor). The proposed method analyzes the relational dependence among the responses of sensors at various nodes as the targets walk through the network. The energy distribution across the network for different number of targets walking at different distances from the nodes has been analyzed to predict the number and location of targets in the sensor network field. The proposed concept has the advantages of having fast execution time and low memory requirements and is potentially well-suited for real-time implementation on *in-situ* computational platforms. Keywords—Personnel detection, seismic sensing, acoustic sensing, sensor-network-based fusion.

9:20

3aSP4. Experimental research on acoustic array sonde in borehole azimuthal reflection logging tool. Xiaolong Hao, Xiaodong Ju, Xiling Wu, Junqiang Lu, Baiyong Men, and Zhijun Yu (State Key Lab. of Petroleum Resources and Prospecting, China Univ. of Petroleum, China University of Petroleum, No. 18 Fuxue Rd., Changping, Beijing, China, haoxl315024@163.com)

Phased combined arc array technology was adopted when designing acoustic sonde to enhance 3D detection capability and resolution in borehole azimuthal reflection logging tool. After constructing experimental measurement system, horizontal and vertical radiation characteristics of the transmitter with different phased excitation delays were tested in a 5 m*5 m*4 m water tank as well as consistency and phased combined effects for 80 array elements in 10 stations of the receiver. Also, this tool was placed in an open waters to explore the response of reflector including size, placed radial distance, and azimuth with optional phased excitation delay. Experimental results show that when the phased excitation delay is 30 us, the main lobe direction of vertical radiation energy of the transmitter could be updeflected by 150 and less side lobe energy. The −3 dB width of main lobe is obviously narrowed using 3 array elements combination in receiver. This tool possesses the ability of recognizing a 2 m*1 m*0.05 m aluminum plate at 10 m radial distance which means about 40 m in real formation, the reflected wave energy intensity distributes symmetrically in eight directions. This work is helpful for acquiring performance of the tool and data processing.

9:35

3aSP5. Research on detection technology for aligned array acoustic sonde of acoustic logging tool. Dong Liu, Xiaodong Ju, Junqiang Lu, Baiyong Men, and Zhijun Yu (China Univ. of Petroleum, No.18, Fuxue Rd., Changping District, Beijing, China, teamo_0911@sina.com)

While aligned array transmitting and receiving acoustic sonde helps in improving the acquisition accuracy and 3D resolution of signals, it also increases the difficulty of detecting and maintaining the sound sonde. A debugging framework based on embedded ARM7&uclinux platform is built; a functional board is designed to simulate the process of transmitting and receiving acoustic sonde, and an evaluation method on the validity and consistency of aligned array probes is proposed. In this design, ARM front terminal is interconnected with host computer through network, and test function board communicates with front terminal through an imitation PC104 bus. The test function board comprises a signal sampling and processing module and an analog signal transmitting module. When debugging the transmitting acoustic sonde, the sampling and processing module with built-in receiver simulates the receiving acoustic sonde, processes the signal by sampling, amplifying, filtering, and analog-digital converting, and then uploads to the host computer. When debugging the receiving acoustic sonde, the analog signal transmitting module, based on DDS technology, simulates periodical sound fields with power horn, which include imitation strata longitudinal wave, shear wave, and stoneley wave. The experiment shows that this design would reduce failure rate of acoustic sonde and improve work efficiency.

9:50–10:05 Break

10:05

3aSP6. A sparse recovery technique for the turbulent boundary layer wavenumber-frequency spectrum with non-uniformly spaced array of pressure transducers. Shaun D. Anderson and Philip W. Gillett (Naval Surface Warfare Ctr. Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, shaun.d.anderson@navy.mil)

Historically, determination of wavenumber-frequency spectra of turbulent wall pressure fluctuations has employed the use of evenly spaced linear arrays of sensors. The reason for this is that even spacing of pressure samples in both the temporal and spatial domains facilitates analysis by direct application of 2-D Fourier transforms. The spacing of pressure measurements directly impacts the resolution of the resulting wavenumber-frequency spectra. This rigid spacing requirements of traditional methods may preclude analysis altogether with even a single sensor failure within the array. Rather than determining sensor locations based on the simplest analysis method, the sparseness of the wavenumber-frequency spectra can be leveraged to allow non-uniform sampling in both temporal and spatial domains. Sparse recovery techniques as an analysis tool for wavenumber-frequency spectra will be demonstrated both theoretically and experimentally for uniformly spaced arrays, uniformly spaced arrays with missing measurements, and randomly spaced arrays.

10:20

3aSP7. Dynamic time warping: Compensating for temperature variations. Alexander C. Douglass (Mech. Eng., Univ. of Utah, 1444 S Beacon Dr., Salt Lake City, UT 84108, u0992315@utah.edu) and Joel B. Harley (Elec. Eng., Univ. of Utah, Salt Lake City, UT)

Guided wave structural health monitoring has the potential to monitor large structural areas. Yet, temperature variations are known to misalign measurement data and cause false alarms. Current popular temperature compensation methods include the optimal signal stretch method and interpolation method. While these methods perform well for small temperature changes, they fail in large temperature variations. In this paper, dynamic time warping is used to realign signal responses that have been distorted by temperature. Baseline subtraction between the realigned signal and the baseline can then be used to detect and identify damage responses. We demonstrate dynamic time warping with guided wave measurements taken from an aluminum plate that is heated in cycles from roughly 75°F to 125°F. Dynamic time warping achieves a correlation between the chosen baseline and the pre-damage measurements greater than 90%. Applying no temperature compensation or more traditional methods, such as optimal signal stretch, demonstrate correlations of 0% and 30%, respectively. This shows that dynamic time warping removes temperature effects more accurately than other current approaches.

10:35

3aSP8. Modifying nearfield acoustic holography for use in building leak detection. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov), Kasanthamy Chelliah, Hirenkumar Patel, and Ganesh Raman (Mech, Mat, and Aero, Engr, Illinois Inst. of Technol., Chicago, IL)

The use of nearfield acoustic holography (NAH) for locating and sizing leaks in building envelops has been previously presented [J. Acoust. Soc. Am. **137**, 2233 (2015); J. Acoust. Soc. Am. **137**, 2325 (2015), and J. Acoust. Soc. Am. **136**, 2172 (2014)]. Standard NAH algorithms are fairly limited in the distance from the sources at which reconstruction can be made. In this paper, the authors present new research that shows that with judicious choices of matrix regularization, the distance from source to NAH measurement can be greatly increased making use of NAH for estimating building leakage much more practical. The application several different NAH matrix regularization methods are compared.

10:50

3aSP9. Method for effective implementation of an acoustic based Hostile Fire Detection System on moving vehicles. Wayne E. Prather (NCPA, Univ. of MS, P.O. Box 1848, 145 Hill Dr., University, MS 38677-1848, wayne@olemiss.edu) and William G. Frazier (Hyperion Technol. Group, Inc., Tupelo, MS)

Due to background noise issues, effective implementation of an acoustic array as part of a Hostile Fire Detection System on moving vehicles has been a significant challenge to date. A new method utilizing both software and hardware design parameters has been developed and successfully demonstrated to be very effective at high speeds on various vehicles. This method has been shown to be effective against both flow based noise sources and acoustic wave based noise sources in low signal to noise ratio conditions. The technology will be discussed and results from measurements utilizing conformally mounted acoustic sensors on moving ground vehicles will be shown.

11:05

3aSP10. Spatial power spectral estimation using coprime sensor array with the min processor. Yang Liu, John R. Buck, and Radienxe Bautista (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, yang.liu@umassd.edu)

A coprime sensor array (CSA) commonly estimates the spatial power spectral density (PSD) of the observed signal by multiplying one conventionally beamformed subarray scanned response with the complex conjugate of the other [Vaidyanathan and Pal, 2011]. This product processor removes the CSA subarray spatial aliasing ambiguities, but has a peak sidelobe higher than the peak sidelobe of a fully populated uniform linear array (ULA) [Adhikari *et al.*, 2014]. Moreover, the resulting spatial PSD estimate is not guaranteed positive semi-definite, and as a result, weak sources can be masked by the negative side lobes of strong interferers. This paper proposes choosing the minimum between the two CSA subarray periodograms at each bearing to resolve the spatial aliasing ambiguities. This min processor achieves lower peak sidelobe height and total sidelobe area than the product processor and preserves the PSD positive semi-definite characteristic. Closed-form expressions for the first two moments of the CSA min PSD estimator are available. Simulations and real data show that the min processor achieves a lower PSD estimate variance than the product processor while keeping the PSD estimate approximately unbiased. [Work supported by ONR BRC Program.]

11:20

3aSP11. Azimuthal evaluation of formations near borehole: Numerical simulations and field examples. Peng Liu, Wenxiao Qiao, Xiaohua Che, Xiaodong Ju, and Junqiang Lu (China Univ. of Petroleum (Beijing), No. 18,Fuxue Rd., Changping District, Beijing, Beijing 102249, China, liupeng198712@126.com)

Azimuthal evaluation of formation properties near borehole is an important research direction of well logging technology, and we develop an innovative 3D sonic logging tool for that purpose. Except conventional measurements of monopole and dipole, this tool is added into an azimuthal transmitter station and a set of azimuthal receiver stations, all of which consist of eight piezoelectric vibrators in circumference to excite or acquire sonic waves. Our numerical simulation results show it can detect the heterogeneity of formations by using the measurement mode of azimuthal transmission and azimuthal reception that two types of P-wave with different velocities are found in the waveforms from some azimuths, but only one P-wave exists in other azimuths. Even two types of S-waves in one waveform are discovered when the formations are very hard. Similar phenomena for both P- and S-waves are also observed in field examples.

WEDNESDAY MORNING, 25 MAY 2016 SALON A, 8:00 A.M. TO 11:50 A.M.

Session 3aUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Sediment Characterization Using Direct and Inverse Techniques I

David P. Knobles, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

3aUW1. An overview of the Sediment Characterization Experiment 2015 survey cruise. Preston S. Wilson and David P. Kmobles (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

The acoustics of fine grained marine sediments, how they impact shallow water acoustic propagation, and the use of inverse and statistical inference techniques for sediment characterization are being studied in this multi-year, multi-institutional project, which is focused on an area of the Southern New England continental shelf known as the New England Mud Patch. An overview of the first survey cruise associated with this project, which took place in July and August of 2015, will be presented. Measurements during that cruise included sub-bottom profiling and multi-beam bathymetry surveys of the experimental area, as well as CTD and gravity core, box core, and multicore sampling. A limited number of acoustic propagation measurements were made along with ship-board laboratory measurements of recovered sediment sound speed and attenuation measurements. A preliminary assessment of the presence and effect of ocean bottom biology was conducted. In this talk, an overview of the survey cruise will be presented. Later in the session, a number of other more detailed presentations on the individual measurements conducted during the survey will appear, along with talks on various other aspects of the acoustics of fine-grained marine sediments, and their impact on underwater acoustics. [Work supported by ONR.]

8:25

3aUW2. Oceanographic constraints on seafloor surveys of the New England continental shelf. Christian de Moustier (10dBx LLC, PO Box 81777, San Diego, CA 92138, cpm@ieee.org)

In July 2015, a seafloor survey of a 30 km x 10 km area of the New England continental shelf, centered at about 40.4750 N, 70.6030 W with an average depth of 74 m, highlighted constraints imposed by the locally dynamic oceanography. Variability in time and space of the water column in the survey area was observed in CTD casts, in underway temperature and salinity data, and in pervasive refraction correction errors symptomatic of internal waves. Depth profiles of temperature and practical salinity show evidence of sporadic continental slope water intrusion on the shelf between 50 m below sea level and the bottom. Underway data reveal a front responsible for an abrupt 10-m/s drop in sound speed along track at 5-m depth. Short of restricting soundings to ± 300 about nadir, there is no mitigation for seafloor surveying through internal waves because frequent sampling of the sound-speed profile along track, e.g., with an underway sound-speed profiling system, provides no information on the water masses on either side of the track, which if different due to internal waves cause refraction correction errors because the assumption of a local horizontally stratified water column is violated. [Work funded by ONR-32OA.]

8:45

3aUW3. Recent oceanographic variability and warming of the continental shelf south of New England and implications for acoustic propagation and sub-bottom characterization. Glen Gawarkiewicz (Woods Hole Oceanographic Inst., M.S. #21, Woods Hole, MA 02543, ggawarkiewicz@whoi.edu)

The continental shelf south of New England is a complex oceanographic environment for underwater acoustics. The combination of large seasonal changes in stratification, complicated frontal phenomena associated with the shelfbreak front, and small spatial scales of variability for soundspeed results in important effects on acoustic propagation and sub-bottom characterization. Two recent trends have also become important. Longer-term warming, on the decadal scale, is affecting shelf water temperatures and soundspeed profiles. In addition, from April 2014 to December 2015, there was a substantial influence on shelf water mass properties from Gulf Stream warm core rings impinging on the continental shelf. The rings also diverted the shelfbreak front onshore at times as well. Some implications for propagation and sub-bottom characterization will be discussed.

9:05

3aUW4. Preliminary characterization of surficial sediment acoustic properties and infauna in the New England Mud Patch. Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Gabriel R. Venegas, Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin, AL), Allen H. Reed (Marine GeoSci. Div., Naval Res. Lab., Tennis Space Ctr., MS), and Ellen Roosen (Wood's Hole Oceanographic Inst., Wood's Hole, MA)

Biology is prevalent on and within many ocean bottom sediments. Organisms can include animals dwelling at or near the water-sediment interface or infauna living within surficial sediments. Bioturbation from burrowing, tube building, or other activities can have physical effects on the sediment acoustic properties. As part of the Sediment Characterization Experiment, a survey cruise was conducted in August 2015 in the New England Mud Patch, a region in the Atlantic continental shelf characterized by a layer of mud up to 12 m thick overlying a sandy subbottom. In addition to gravity coring operations to determine the properties of the mud layer, box cores and multicores were collected to examine the surficial sediment properties. Infauna were prevalent in the surficial sediment samples and were collected and characterized for body size, hardness, and potential for bioturbation or structuring. Shipboard measurements of shear and compressional waves were performed on the box core samples using time-of-flight measurements. Preliminary results from the infauna analysis and the shipboard acoustic measurements will be presented. [Work supported by ONR.]

Contributed Papers

9:25

3aUW5. Shipboard low frequency sound speed measurements in the New England Mud Patch. Gabriel R. Venegas, Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu), Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs., Austin, TX), Allen H. Reed (Marine GeoSci. Div., Naval Res. Lab., Austin, Texas), Ellen Roosen (Geology and Geophys. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Daniel Ecker, Kyle Fringer, and Joseph P. Smith (Oceanogr. Dept., U. S. Naval Acad., Annapolis, MD)

Muddy ocean bottoms have been shown to be acoustically softer than the water column, presenting challenges in resolving inverse problems such as in remote parameter sensing and target detection. In addition, the acoustic properties of mud are less adequately understood compared to sandy sediments. As part of the Sediment Characterization Experiment, a survey cruise was conducted in August 2015 in the New England Mud Patch, in order to further study the acoustic properties of mud. In addition to gravity coring operations to determine the properties of the mud layer, box cores and multicores were collected to examine the surficial sediment properties. Samples were extracted from within the multicores and transferred to a resonator tube, where shipboard low frequency laboratory measurements of sound speed were performed. Before discarding the sample, it was perturbed and allowed to settle in the resonator over a period of two days, during which low frequency sound speed measurements were taken. Experimental challenges and preliminary results from the acoustic measurements and settling experiment will be presented. [Work supported by ONR.]

9:40

3aUW6. Estimation of sound speed and attenuation in mud sediments using combustive sound source signals measured on the New England continental shelf. Lin Wan, Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 104 Robinson Hall, 261 S. College Ave., Newark, DE 19716, wan@udel.edu), David Knobles (Knobles Sci. and Anal., LLC, Austin, TX), Preston Wilson (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and John Goff (Inst. for Geophys., Univ. of Texas at Austin, Austin, TX)

Acoustical measurements were made in coincidence with an environmental survey for future sediment characterization experiments in the mud patch region of the New England continental shelf from July 22 to August 3, 2015. One vertical line array (VLA) containing six hydrophones and 15

environmental sensors was deployed in 75-m water at the center of the survey (40.477°, −70.604°) in order to record noise data and the acoustic signals generated by the combustive sound source (CSS) [McNeese *et al*., J. Acoust. Soc. Am., 2014]. Fifteen CSS shots detonated at various ranges (2.5 km and 14.5 km) and depths (5 m, 10 m, and 20 m) were recorded by the VLA. The sub-bottom layering structure with a mud layer overlaying a sand bottom was obtained from the CHIRP sonar survey. This paper utilizes the modal dispersion characteristics to invert the sound speed in the bottom. Then, the sound attenuation in mud as a function of frequency is estimated using the transmission loss (TL) data. Measured modal dispersion curves are compared with modeled dispersion curves based on the inverted bottom acoustic parameters. [Work supported by ONR.]

9:55–10:10 Break

Invited Papers

10:10

3aUW7. Suggested methodology for direct and inverse characterization of marine sediments that contain mud. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Mud consists mostly of particles of clay minerals: kaolinite, illite, and smectite. Particles are hexagonally shaped platelets and have widths of the order of microns. In contrast, sandy/silty sediments are composed of silicate particles that are spheroidal is shape and have sizes of the order of millimeters. One distinction is that the platelets commonly carry electrical charges that repel each other. Relevant physics underlying their acoustic properties is considerably different than that for sandy/silty sediments. Naturally occurring mud also contains sandy/silty content, biological matter, and gaseous bubbles. The present paper discusses various techniques by which one can estimate quantitative information about mud sediments. Much can be learned from detailed geochemical analysis and microscopic observation of core samples, but enough is known about mud's general properties so that indirect methods can be used. These can estimate the density and sound speed as functions of depth into the sediment. The sound speed in mud is better approximated by the Mallock-Wood equations than for sandy/silty sediments. Density gradients in conjunction with statistical mechanics considerations yield information on how much of the mud is clay. Measurement of compressional wave attenuation can give information about the forces between platelets. [Work supported by ONR.]

10:30

3aUW8. Geoacoustic inference and the search for ground truth. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Jan Dettmer, and Stan D. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

A variety of commercial, military, and scientific applications require knowledge of seabed properties. In the ocean acoustics community, the properties typically of interest are sound speed, density, and attenuation and sometimes shear speed and attenuation. Numerous approaches have been developed to estimate these by geoacoustic inference, i.e., by measuring a quantity (e.g., reflection coefficient, transmission loss, or pressure across a hydrophone array) and estimating the seabed properties from the data. Inference requires a large number of assumptions on the depth, range, and frequency dependencies of ocean and seabed properties. These assumptions are widely, and often necessarily, made with minimal supporting information. In cases where the actual physical seabed properties are of interest there is an important requirement to validate the result. Measurements on sediment cores are widely called "ground truth." However, these data frequently contain bias errors and exhibit rather large uncertainties, sometimes larger than those from geoacoustic inference. Here, difficulties and opportunities associated with collecting, conducting, and interpreting measurements on cores are discussed. Cores can be a useful independent measurement of sediment properties, but should not be termed as "ground truth." [Work supported by the Office of Naval Research and the Centre for Maritime Research and Experimentation.]

10:50

3aUW9. Marine sediment core collection methods and analytical techniques for shallow water sub-seafloor acoustic characterization. Jason D. Chaytor (U.S. Geological Survey, 384 Woods Hole Rd., Woods Hole, MA 02543, jchaytor@usgs.gov)

Collection of intact sub-seafloor sediment samples from aqueous environments requires a suite of coring and drilling tools that vary in complexity, size, and cost as a function of the location, thickness, and analysis requirements of the sediment to be recovered. In shallow water, continental shelf environments such as the Mud Patch offshore southern New England, piston-, gravity-, and vibra-coring techniques allow for the collection of continuous sediment cores that can exceed 10 m in length through a variety of fine- and coarse-grained sediments. Piston and gravity corers, which penetrate the seafloor through a combination of system weight and velocity, operate most effectively in clay and silt dominated materials, recovering relatively undisturbed cores suitable for a wide range of sedimentological, geotechnical, and acoustical analyses. Recovered cores proceed through a series of non-destructive and destructive testing stages to fully characterize the sediments, beginning with physical properties logging (e.g., wet bulk density, p-wave velocity, and magnetic susceptibility), visual stratigraphic description, shear-wave velocity, and photographic and x-ray/CT imaging. Laboratory testing of extracted sub-samples includes, but is not limited to, geotechnical index properties (e.g., water content, undrained shear strength, and density), grain size analysis, carbonate and organic material concentrations, and mineral content (via x-ray diffraction).

11:10

3aUW10. Estimation of shear wave properties using Scholte wave inversions. Gopu R. Potty and James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu)

Shear speeds in semi-consolidated and consolidated shallow water sediments can significantly impact compressional wave attenuation and arrival times of acoustic normal modes. One of the most promising approaches to estimate shear speed is to invert the shear speed profile using the dispersion of interface waves (Scholte waves). The propagation speed of the Scholte waves is closely related to the shear wave speed over a depth of 1–2 wavelengths into the seabed. A geophone system for the measurement of these interface waves, along with an inversion scheme that inverts the Scholte wave dispersion data for sediment shear speed profiles and shear attenuation has been developed. A forward model based on the dynamic stiffness method has been developed and implemented in the inversion. A new Interface Wave Sediment Profiler (iWaSP) is being currently developed for the measurement of bottom properties such as shear speed and attenuation in the top 1–2 m of a variety of sediment types (including mud) with a wideband, vibratory source, and accelerometers with bandwidth up to 1 kHz. Inversions using data from previous deployments will be presented and proposed experimental designs for the proposed seabed experiment will be discussed. [Work supported by Office of Naval Research.]

11:30

3aUW11. *In situ* measurements of sediment sound speed and attenuation in the frequency band of 0.5–10 kilohertz. Jie Yang (Acoust. Dept., APL-UW, 1013 NE 40th St., Seattle, WA 98105, jieyang@apl.washington.edu)

Knowledge of sediment sound speed and attenuation is crucial for predicting sound propagation. The sediment Acoustic-speed Measurement System (SAMS) was designed to measure frequency dependence of sound speed and attenuation within the surficial 3 m in the frequency band of 2–50 kHz. SAMS has two independent and interchangeable drill systems: one employs suction and the other a water jet. The suction system has minimal disturbance on the medium around the penetrating probe while the water jet system can help penetrate consolidated shell/sand layers. Depending on sediment types encountered, the two systems can be interchanged on site in less than 4 h. SAMS was successfully deployed during the Shallow Water 2006 experiment on the New Jersey continental shelf. It was also deployed during the Gulf Experiments 2011, 2012, and the Target and Reverberation Experiment 2013, all conducted off Panama City, FL, where sediment types range from mud to coarse sand. Sediment sound speed and attenuation results from the two experiments are presented. In preparation for the upcoming Seabed Characterization Experiment, the effort to extend measurements down to 500–1500 Hz, where dispersion is of particular importance, will be discussed as well. [Work supported by ONR.]

WEDNESDAY AFTERNOON, 25 MAY 2016 SALON B/C, 1:15 P.M. TO 3:00 P.M.

Session 3pAA

Architectural Acoustics: Relating Perception to Room Acoustics Measurements and Metrics in Performing Arts Venues II

Michelle C. Vigeant, Cochair
Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Gregory A. Miller, Cochair
Threshold Acoustics, LLC, 53 W. Jackson Boulevard, Suite 815, Chicago, IL 60604

Chair's Introduction—1:15

Invited Papers

1:20

3pAA1. Comparison of assorted diffusive room conditions. Jay Bliefnick and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, jbliefnick@huskers.unl.edu)

A number of objective metrics have been proposed to quantify diffusive conditions in rooms, but less has been discovered about how perception of diffusion links to those metrics. In this project room impulse responses from a range of diffusive room conditions were recorded in the Mocap variable acoustics facility at Columbia College Chicago, which features 1200 sq. ft. of reversible absorptive/diffusive panels. Modifications to these surfaces were implemented to conduct two tests under multiple wall configurations. The first test concentrated on a direct reflection off of a 16 ft. x 8 ft. test wall, which was progressively changed from fully absorptive to fully

diffusive. The second test utilized the entire facility by arranging the diffusers in multiple coverage percentages (10%–100%) and in three unique arrangements. A number of metrics have been calculated from the gathered impulse responses and compared, including those proposed to account for room diffusivity. Measured binaural room impulse responses are being used to generate auralizations for use in testing to discern when changes in the sound field become perceptible. Results to date will be presented.

1:40

3pAA2. Design consideration of sound diffusers for wall surfaces in concert halls. Jin Y. Jeon, Hyung S. Jang, and Hansol Lim (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, South Korea, jyjeon@hanyang.ac.kr)

A wall diffuser has been designed for high scattering and diffusion coefficients using tenth-scale models. The diffuser profiles of the diffuser were designed based on using the Voronoi diagrams to create a randomized pattern with a plane surface with a particular structural height. The intaglio and embossed patterns of these profiles were compared by measuring the scattering and the diffusion coefficients. The intaglio diffusers showed higher scattering and diffusion coefficients than the embossed diffusers. Moreover, random arrays of the profiles presented resulted in higher scattering and diffusion coefficients than the sequential arrays ordered sequentially in terms of the profile in the structural height. The effects of diffusers on sound-field diffuseness were also investigated, and it was found that the diffusive surfaces tend to produce more diffused early reflections than the fine reflective surfaces.

2:00

3pAA3. Comparisons between room acoustics preferences of musicians and non-musicians regarding solo-instrument and orchestral motifs in auralizations. Martin S. Lawless (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu) and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., University Park, New York)

Musicians are good candidates for room acoustics subjective testing since they regularly practice critical listening skills, but recent work suggests that non-musicians should also be included as the two groups may have diverse room acoustics preferences. A study was conducted to evaluate these two participant groups to ensure the representation of the entire population in room acoustics studies. The specific goal of the present work was to investigate preference differences between musicians and non-musicians in the evaluation of musical stimuli with varying reverberation time (RT). Preliminary data introduced at a prior meeting demonstrated that on average musicians and non-musicians displayed similar preference trends for auralizations of two solo-instrument motifs. However, a k-means clustering analysis revealed that a greater percentage of non-musicians exhibited statistically indistinct preference ratings across the stimuli, denoted as the "no preference" group. This result may indicate that non-musicians are not good candidates for room acoustics testing. The present study expands upon the preliminary work with additional participants that assessed both the solo-instrument motifs and two new orchestral motifs simulated in a concert hall with RT ranging from 0.0 to 7.2 s. The preference trends were also compared to the types of musicians tested.

2:20

3pAA4. Acoustical performance of a completely renovated music recital hall from subjective and objective points-of-view. Abigail Davis and Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, a849d414@ku.edu)

Swarthout Recital Hall located in Murphy Hall at the University of Kansas was initially constructed in the 1950s, with the original interior remaining largely intact until extensive renovations were completed in 2015. Two major complaints regarding the original hall were a lack of feedback from the hall to the performers onstage, and the acoustical coldness of the hall. With excellent cooperation between the School of Music Dean and the music faculty, the project architect, and the acoustic consultant and through the use of computer modeling, a new interior envelope was designed and interior materials selected which addressed these issues. In response, the renovated hall has received highly positive feedback from performers and patrons in both of these respects. This paper will address the measurable acoustic parameters of the renovated hall and how they correlate to both the digitally modeled versions of the hall and the hall acoustic perceptions by performers and audience.

2:40

3pAA5. Real & virtual: Lessons learned from modeling, measuring, and listening in a multi-purpose hall. William Chu and Brandon Cudequest (McKay Conant Hoover, 5655 LIindero Canyon Rd., Ste#325, Westlake Village, CA 91362, wchu@mchinc.com)

McKay Conant Hoover has guided a 2300-seat multipurpose venue through a series of modest renovations, each with tangible improvements, including expansion of the orchestra pit, provision for a proscenium eyebrow, reconfiguration and replacement of the orchestra shell, HVAC noise reduction, sidewall shaping, and finish selections to improve hall response for unamplified acoustics. However, much of the distinctive geometry of the hall by the renowned acoustician, Vern Knudsen, such as the broad, curved ceilings and walls, and the arching flying balcony, has remained unchanged since its opening in 1972. In 2014, an opportunity arose to investigate the acoustics of this space through a series of critical listening exercises, *in-situ* impulse response measurements, and virtual 3-D CATT modeling and auralizations. This paper will explore the challenges of comparing measured data, subjective preferences from actual music performances, computational model results, and listening tests through auralizations.

Session 3pAB

Animal Bioacoustics: Airborne/Automatic Animal Bioacoustics

Rolf Mueller, Chair
Mechanical Engineering, Virginia Tech, ICTAS Life Sciences District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061

Contributed Papers

1:30

3pAB1. Asymmetric multi-frequency biosonar beam pattern of tongue-clicking bat, *Rousettus aegyptiacus*. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), Benjamin Falk, Chen Chiu, Anand Krishnan, and Cynthia F. Moss (Dept. of Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

The beam pattern of sonar signals emitted by echolocating animals, such as bats and toothed whales, directly influences the acoustic information available for guiding task-specific behaviors. The lingual echolocating bat, *Rousettus aegyptiacus*, emits broadband transient sonar clicks that resemble those of dolphins. The clicks are emitted in left-right pairs, with the maximum intensity slope of each signal pointing toward the target during navigation. However, detailed beam pattern characteristics of these lingual sonar clicks remain unknown. Using a loosely populated three-dimensional microphone array, we systematically characterize the multi-frequency beam structure of *R. aegyptiacus* in the entire azimuth-elevation domain. The bat's head aim was recorded by an infrared high-speed motion-capture camera system. We show that the sonar beam of *R. aegyptiacus* tongue clicks is vertically elongated and exhibits an unusual multi-frequency structure that has not been described previously in the literature. Specifically, the high-intensity portion of the beam shifts medial in azimuth from the left and right click directions and downward in elevation with increasing frequency. Combined with close-up videos of mouth movement as the bats click, these results suggest a more sophisticated beam formation and steering mechanism than conventional simple aperture models.

1:45

3pAB2. A biomimetic perspective on the dynamics in horseshoe bat biosonar. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Tech, Blacksburg, VA), Philip Caspers (Mech. Eng., Virginia Tech, Blacksburg, VA), and Yanqing Fu (Biomedical Eng. and Mech., Virginia Tech, Blacksburg, VA)

Research with behaving horseshoe bats has suggested that the animals' biosonar system is characterized by pervasive dynamics at the interfaces for ultrasound emission and reception. Baffle shapes that diffract the outgoing and incoming ultrasonic pulses change their geometries during emission and reception. However, it remains unclear if and how this dynamics could affect the function of bats' biosonar system. To investigate this question, biomimetic reproduction of this peripheral dynamics have been used to obtain system characterizations as well as echo recordings. The data obtained with these biomimetic sonar systems have yielded the following key results: If the static and dynamic geometries of the diffracting baffles as well as their coupling to the transducers is suitable, time-variant device characteristics can be produced—even for small and geometrically simple changes to the shapes. These characteristics depend on direction and frequency (like a static beampattern) as well as on time. Furthermore, the time-variant device characteristics were found to result in likewise time-variant signatures imposed on echoes from targets with simple geometries as well as natural targets such as foliages. It remains to be investigated whether and how horseshoe bats could make use of these time-variant echo signatures for the encoding of sensory information.

2:00

3pAB3. Quantifying the noseleaf and pulse dynamics in rhinolophid and hipposiderid bats. Luhui Yang, Liujun Zhang, Ru Zhang (Shandong Univ. - Virginia Tech Int.. Lab., Shandong Univ., Shanda South Rd. 27, Jinan, Shandong 250100, China, 913022794@qq.com), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Horseshoe bats (Rhinolophidae) and Old-World leaf-nosed bats (Hipposideridae) are two related bat families that emit their biosonar pulses nasally and diffract the outgoing wave-packets with elaborate baffle shapes ("noseleaves"). Since the noseleaf surfaces are frequently in motion during pulse emission, an experimental setup has been established to characterize the dynamics in their geometry in conjunction with the effects that this dynamics may have on the ultrasonic pulses. To achieve this goal, greater horseshoe bats (*Rhinolophus ferrumequinum*), great roundleaf bats (*Hipposideros armiger*), and Pratt's roundleaf bats (*Hipposideros pratti*) were trained to emit biosonar pulses while seated on a platform. At least 10 landmarks were placed on the noseleaves of the animals to track the dynamic geometry of these structures with an array of four high-speed video cameras (frame rate 400 Hz). Pairs of video frames were used to reconstruct the 3d trajectories of the noseleaf landmarks, and from these trajectories, shape changes were assessed based on point-distances and measures derived from them. The emitted ultrasonic pulses were measured as a function of direction with a cross-shaped array of ultrasonic microphones. The dynamics in pulse waveforms that were emitted concurrent with noseleaf motions was assessed using correlation and variability measures within and across channels.

2:15

3pAB4. Automatic fish sounds classification. Marielle Malfante, Mauro Dalla Mura, Jerome I. Mars (GIPSA-Lab, UGA, 11 rue des Mathematiques, Grenoble 38402, France, marielle.malfante@gipsa-lab.grenoble-inp.fr), and Cedric Gervaise (Chaire CHORUS, Fondation Grenoble-INP, Grenoble, France)

The context of this work is environmental monitoring. Specifically, we focus on acoustic systems for monitoring fish populations. By investigating the recorded sounds of fishes, it is possible to monitor the spatial and temporal evolution of fish populations. The aim of this work is to present an automatic fish sounds classification system. The solution we propose is new and based upon supervised machine learning (in particular, classification is performed by Random Forest). The features used in input of the learning algorithm come from an extensive state of the art in various domains of classification such as speech, music, animal calls, environmental acoustic landscape, and human induced noises. Our system is trained and tested on nighttime recordings from a 20 m depth seagrass habitat (Calvi, Corsica, France). From this study, we propose to consider 66 different features

3p WED. PM

(shape and/or statistical description in time and frequency). Fish sounds are automatically classified into four different classes (drums, grunt, impulse, and FM), and our system reaches 94% of correct classification rate compared to 77% when considering MFFC features. In order to deal with large datasets and to study the evolution of fish populations, we are currently developing an approach based on dynamic classification with rejection (negative class).

2:30

3pAB5. Animal aeroacoustics: Fluttering feathers and humming hummingbird wings. Christopher J. Clark and Emily Mistick (Dept. of Biology, Univ. of California Riverside, Riverside, CA 92521, cclark@ucr.edu)

Hummingbirds are so-named for the humming sounds their wings produce in flight. We review prior research on aerodynamic mechanisms that produce hummingbird wing sounds, such as aeroelastic flutter. We then present a new aeroacoustic model of the humming of hummingbird wings. This model was tested over a range of flight speeds on hummingbirds flying in an acoustic wind tunnel, and filming them with an Optinav acoustic camera. Possible implications of this model for the sounds produced by micro-air vehicles ("drones") are discussed.

WEDNESDAY AFTERNOON, 25 MAY 2016 SALON H, 1:00 P.M. TO 2:55 P.M.

Session 3pBA

Biomedical Acoustics: Focused Ultrasound for Brain Treatments

Nathan McDannold, Cochair
Radiology, Brigham and Women, 75 Francis St., Boston, MA

Kullervo Hynynen, Cochair
Medical Biophysics, University of Toronto, Sunnybrook Health Sciences Centre, Toronto, ON M4N 3M5, Canada

Chair's Introduction—1:00

Invited Papers

1:05

3pBA1. Acoustically controlled ultrasound-mediated drug delivery to the central nervous system. Meaghan A. O'Reilly, Ryan M. Jones, Olivia Hough, and Kullervo Hynynen (Sunnybrook Res. Inst., 2075 Bayview Ave., Rm. C713, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

The barriers of the central nervous system (CNS), including the blood-brain barrier (BBB) and blood-spinal cord barrier ((BSCB), prevent most drugs from reaching therapeutically significant concentrations in these tissues. Thus, the BBB and BSCB are significant obstacles in the treatment of CNS disorders. When combined with microbubbles, ultrasound can be used to transiently open these barriers to facilitate the delivery of drug, gene and cell therapies. There exists a regime of microbubble excitation where a completely reversible opening effect can be achieved without damaging the tissue. We, and others, have previously presented methods to control the acoustic exposures during treatment to target this regime, and thus ensure efficacy and safety. This talk will present recent work in ultrasound-mediated drug delivery to the CNS. New findings of the role of microbubble emissions in controlling these exposures will be discussed, including dose-dependency and the temporal characteristics of stable cavitation signatures.

1:25

3pBA2. Transcranial thermal ablation with a 230 kHz MRI-guided focused ultrasound system in a large animal mode. Nathan McDannold, Jonathan Sutton, Natalia Vykhodtseva (Radiology, Brigham and Women, 75 Francis St., Boston, MA, njm@bwh.harvard.edu), and Margaret Livingstone (Neurobiology, Harvard Med. School, Boston, MA)

We evaluated the feasibility of thermal ablation in using a 230 kHz transcranial MRI-guided Focused Ultrasound (TcMRgFUS) system in three rhesus macaques. Thalamic targets were sonicated at 40–50s at 90–560 acoustic Watts. Focal heating sufficient to create an MRI-evident lesion was achieved in 4/6 targets where thermal dose exceeded 240 CEM43°C. Focal heating increased linearly as a function of the applied energy at a rate of 3.2 ± 0.4°C/kJ (R^2:0.81). Lesion sizes were consistent with 240 CEM43°C contours. The findings suggest that the lesions were consistent with thermal mechanisms. No evidence of cavitation-related petechiae were evident after sonication. Similar tests in macaques with a version of this system operating at 670 kHz (Hynynen *et al.*, Eur. J. Radiol. 2006) measured skull-induced heating of 130°C/kJ/cm^2 of outer skull surface, more than twice of that measured here (63°C per kJ/cm^2). While no or minimal focal heating was observed at 670 kHz, we reached ablation-level thermal dose values at 230 kHz. Thus, these preliminary results thus suggest that this low frequency system can expand the "treatment envelope" for TcMRgFUS.

1:45

3pBA3. Neuroprotection and neurorestoration through the opened blood-brain barrier using focused ultrasound and microbubbles. Elisa Konofagou (Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

After cancer and heart disease, neurodegenerative diseases, such as Alzheimer's, Parkinson's, multiple sclerosis (MS) and amyotrophic lateral sclerosis (ALS), take more lives each year than any other illness. Although some effective treatments are available, most of those diseases remain undertreated. The blood-brain barrier (BBB) has been proclaimed as the "bottleneck" of brain drug delivery. Focused ultrasound (FUS) in conjunction with microbubbles is currently the only technique that can open the BBB locally, transiently, and noninvasively. Our group has demonstrated that pharmacological compounds of variable sizes can not only traverse the BBB but also induce therapeutic effects in the presence of neurodegenerative disease. In this paper, the trans-BBB neurotherapeutic delivery with FUS is shown to **(1)** be safely facilitated within a specific pressure range in both mice and non-human primates through the mechanism of cavitation, **(2)** be controlled through the pressure amplitude and microbubble characteristics, **(3)** induce downstream signaling effects in the neuronal cell, **(4)** trigger neuroprotection and neurorestoration relevant in the treatment of neurodegeneration, **(5)** be localized and monitored in real time through cavitation monitoring and mapping. BBB opening is combined with systemic administration of neurotrophic molecules or adeno-associated viruses that allow for neuronal protection and growth in the BBB-opened regions.

2:05

3pBA4. Performance of a simulation-based phase aberration correction technique in transcranial ultrasound modeling. Douglas Christensen (BioEng., Univ. of Utah, 50 So Central Campus Dr., Salt Lake City, UT 84112, christen@ee.utah.edu), Dennis Parker (Utah Ctr. for Adv. Imaging Res., Univ. of Utah, Salt Lake City, UT), and Scott Almquist (School of Computing, Univ. of Utah, Salt Lake City, UT)

Due to its irregular geometry and acoustic properties, the presence of the skull in transcranial focused ultrasound therapies can lead to considerable beam phase aberration, resulting in distorted focal spots. There are several possible techniques for correcting this phase aberration with a phased-array transducer, the most accurate of which (although invasive) is hydrophone-based time reversal. There are also simulation-based approaches, and here we exploit the time-advantages of the Hybrid Angular Spectrum (HAS) numerical method and apply it to two models of the skull: a 3D-printed plastic skull analog and a segment of a human skull for which a CT scan has been obtained. We compare the focused pressure patterns—obtained via hydrophone scans—for the cases of no phase correction, hydrophone-based phase correction, and simulation-based (HAS) phase correction. We show the degree to which the correction methods improve the focal spot quality, pressure amplitude, and localization accuracy. The very good results for the analog model compared to results for the skull segment indicate that better mapping of CT Hounsfield units to acoustic properties of the bone is needed. In addition, we use the simulation technique to predict the effects of beam steering and transducer/model misregistration on the effectiveness of aberration correction.

Contributed Papers

2:25

3pBA5. Computation of ultrasonic pressure fields in feline brain. Nazanin Omidi, Cecille Labuda, and Charles C. Church (Dept. of Phys. and Astronomy and National Ctr. for Physical Acoust., The Univ. of MS, 145 Hill Dr., P.O. Box 1848, University, MS 38677-1848, nomidi@go.olemiss.edu)

In 1975, Dunn *et al*. (JASA **58**, 512–514) showed that a simple relation describes the ultrasonic threshold for cavitation-induced changes in the mammalian brain. The thresholds for tissue damage were estimated for a variety of acoustic parameters in exposed feline brain. The goal of this study was to improve the estimates for acoustic pressures and intensities present *in vivo* during experimental exposures rather than estimating them using linear theory. In our current project, the acoustic pressure waveforms produced in the brains of anesthetized felines were numerically simulated for a spherically focused, nominally f1-transducer (focal length = 13 cm) at increasing values of the source pressure at frequencies of 1, 3, and 9 MHz. The corresponding focal intensities were correlated with the experimental data of Dunn *et al*. The focal pressure waveforms were also computed at the location of the true maximum. For low source pressures, the computed waveforms were the same as those determined using linear theory, and the focal intensities matched experimentally determined values. For higher source pressures, the focal pressure waveforms became increasingly distorted; similar results were obtained with increasing frequency.

2:40

3pBA6. Phased array techniques for multiple focus synthesis in transcranial focused ultrasound. Alec Hughes and Kullervo Hynynen (Dept. of Medical Biophys., Univ. of Toronto, 101 College St., Rm. 15-701, Toronto, ON M5G 1L7, Canada, ahughes@sri.utoronto.ca)

Recent clinical successes of transcranial focused ultrasound have occurred in the treatments of essential tremor, neuropathic pain, and Parkinson's disease, among others. We will present results of an investigation into the synthesis of multiple foci using iterative steering through multiple points and phased array controls for multiple focus acoustic patterns. In this numerical study, exported computed tomography (CT) imaging data of the skull was segmented and positioned inside a hemispherical phased array. A combination of full-wave and ray acoustic models were used to simulate the calculation of phased array controls and the resultant acoustic field. Using techniques from previous work on simultaneous multiple focus synthesis and rapidly steered foci in homogeneous media, it is shown that it is possible to elevate the temperature in the brain to therapeutic hyperthermia levels. In addition, potential applications for microbubble-mediated therapies using these techniques are discussed. These results indicate that transcranial hyperthermia over large volumes using focused ultrasound is possible and may have applications to future thermal therapies.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Frederick J. Gallun, Chair
National Center for Rehabilitative Auditory Research, VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239

Chair's Introduction—1:30

Invited Papers

1:35

3pID1. Acoustics on YouTube: Using the internet to improve and inspire learning. Michael B. Wilson (Phys., North Carolina State Univ., 247 RidDC Hall, Raleigh, NC 27695, mbwilson@ncsu.edu)

More people have access to more information than ever before, and programs are organizing and providing educational content for free to millions of internet users worldwide. Popular video, blog, and social media websites have creators devoted to engaging an audience through education. This content ranges from interesting facts and demonstrations that introduce a topic to entire university courses. While styles and target audiences vary greatly, the focus is education, clarifying misconceptions, and sparking an interest in learning. Presented will be a survey of current online education, resources, and outreach, as well as ways the acoustics community can get involved in improving education in and out of the classroom.

1:55

3pID2. The state of the art in worship space acoustics. David T. Bradley (Phys. + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604, dabradley@vassar.edu), Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, Omaha, NE), and Lauren M. Ronsse (Audio Arts & Acoust. Dept., Columbia College Chicago, Chicago, IL)

Thoughtful acoustic design is essential for creating a worship experience full of awe and wonder. Worship space acousticians must grapple with differing acoustic requirements for speech and music, including a diversity of typologies across religions, cultures, and regions. Even a single worship space may require multi-functionality if used for a variety of service types or events. In addition to supporting the speech and/or music delivered as part of a service, the space must be supportive of congregant participation—ranging from lively exaltation to quiet reflection. The result is a richly complex and diverse set of acoustic approaches in the realm of worship space design. The new book, *Worship Space Acoustics: 3 Decades of Design* (Eds. Bradley, Ryherd, + Ronsse, Springer/ASA Press, 2016) provides a wide-ranging tour through churches, synagogues, mosques, and other worship spaces designed during the past 30 years. Acoustical consulting firms, architects, and worship space designers from across the world contributed their recent innovative works, including detailed renderings and architectural drawings, acoustic data graphs, and space descriptions. The content paints a picture of key worship space acoustic design goals and strategies from the last three decades. Case studies and overall themes from the book will be presented.

2:15

3pID3. An example of dissociation between speech intelligibility and perceived reverberation. Pavel Zahorik and Gregory M. Ellis (Dept. of Otolaryngol. and Communicative Disord. & Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

It is well known that reverberation can affect the intelligibility of speech. Psychophysical and computational results have demonstrated that the relationship is inverse: an increase in the amount of reverberation results in a decrease in intelligibility. From the architectural acoustics literature, it is also well known that there is a direct relationship between the physical amount of reverberation and perceived reverberation. It therefore might be assumed that perceived reverberation and intelligibility are inversely related, although here a situation is demonstrated in which the two are effectively independent of one another. Using virtual auditory space techniques to simulate reverberant sound field listening, it is shown that when reverberant sound level is artificially decreased in the ipsilateral ear and naturally preserved in the contralateral ear, perceived reverberation is unaffected, but speech intelligibility is markedly improved. This dissociation likely results from the differential monaural and binaural aspects of reverberation, and is consistent with the idea that perceived reverberation is multidimensional. These results also suggest a potential binaural approach to the application of improving speech intelligibility in reverberation that does not limit the positive sound quality benefits of reverberation.

2:35

3pID4. Detailed measurements on lingual organ pipes for developing innovative methods and software for the pipe design. Judit Angster (Acoust., Fraunhofer IBP, Nobel Str. 12., Stuttgart, 70569, Germany, Judit.Angster@ibp.fraunhofer.de), Peter Rucz (Dept. of Networked Systems and Services, Budapest Univ. of Technol. and Economics, Budapest, Hungary), and Andras Miklos (Steinbeis Transfer Ctr. of Appl. Acoust., Stuttgart, Germany)

Within the framework of a European research project exciting new results have been achieved recently in the research of lingual organ pipes. The main objectives of the project are to solve practical problems of the dimensioning of reed organ pipes, to develop innovative methods and software for helping the sound design work of organ builder enterprises. A better understanding of the role of the shallot and resonator on the attack and the timbre of reed pipes is necessary for this reason. Visualization of reed motion by high speed camera and measurements of reed velocity, wind pressure in the boot and shallot, and sound pressure in the shallot and at the end of the resonator have revealed interesting details about the very complex process of sound generation of lingual organ pipes. These experiments have been carried out in close cooperation with the participating organ builder firms. As examples, a few results of the measurements of Crumhorn and Vox Humana pipes are presented. These results, combined with computer simulations of resonators, can be used for optimizing shallots and resonators. Moreover, the extended knowledge of sound generation can serve as a physical model of sound synthesis.

WEDNESDAY AFTERNOON, 25 MAY 2016 SALON J, 1:15 P.M. TO 2:45 P.M.

Session 3pPA

Physical Acoustics: Atmospheric Acoustic Phenomena II

JohnPaul R. Abbott, Cochair
Department of Physics and Astronomy, National Center for Physical Acoustics, University of Mississippi, 1 Coliseum Dr., Room 1044, Oxford, MS 38677

Gregory Lyons, Cochair
Aeroacoustics, National Center for Physical Acoustics, University, MS 38677-1848

3p WED. PM

Contributed Papers

1:15

3pPA1. Sound transmission measurements through porous screens. Sarah M. Young, Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., BYU, N283 ESC, Provo, UT 84602, sarahmyoung24@gmail.com), Nicholas B. Morrill (Precision Membranes, Provo, UT), Robert C. Davis, and Richard R. Vanfleet (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The two microphone transfer function technique is used to measure sound transmission properties of porous screens or membranes in a plane wave tube. This presentation will compare sound transmission of porous screens from several manufacturers. Measurements are made with two different plane wave tubes, one of diameter 10.2 cm to measure frequencies between 100 Hz and 2 kHz, and the other of diameter 1.3 cm to measure frequencies between 2 kHz and 16 kHz. Special considerations are made to account for the intrinsic losses in the smaller diameter tube.

1:30

3pPA2. A highly portable, lightweight windscreen for infrasound sensors. William G. Frazier (Hyperion Technol. Group, Inc., 3248 West Jackson St., Tupelo, MS 38804, gfrazier@hyperiontg.com) and David Harris (Hyperion Technol. Group, Inc., Tueplo, MS)

It is often desired to deploy infrasound sensors in remote locations for short periods of time. Because of noise created by wind flow past the sensor in the atmosphere at infrasound frequencies, it is also desirable to employ a mechanical windscreen with the goal of reducing the effect of the wind so as to enhance the acoustic signal-to-noise ratio. One of the challenges in wind screen design for temporary applications is to achieve reasonable portability in addition to adequate wind noise filtering. In this presentation, we report the performance a new windscreen design that is highly portable and that achieves performance on par with a commonly used dome-shaped windscreen. Performance comparisons of this newly designed windscreen, a typical porous hose windscreen, and a commonly used dome windscreen are investigated. Specifically, wind noise reduction of these three windscreens compared with a un-screened sensor are presented as a function of frequency and wavenumber using the measured wind speed as a proxy for wave speed. Additionally, acoustic signal noise attenuation is also examined so as to compare signal-to-noise ratio gains. This investigation is similar to the one presented by Alberts, Ludwig, and Talmadge (J. Acoust. Soc. Am. **137**, 2261 (2015)) for other small windscreens.

1:45

3pPA3. Using the Matérn covariance function to enhance detection and estimation of transient infrasound. William G. Frazier (Hyperion Technol. Group, Inc., 3248 West Jackson St., Tupelo, MS 38804, gfrazier@hyperiontg.com)

Adaptive filtering techniques are well known to enhance detection of anomolous events in the presence of slowly changing (quasi-stationary), autocorrelated noise backgrounds such as those occurring in measurements of infrasound signals in the atmosphere. The noise in this case is dominated by the effect of the turbulent wind blowing over the sensing element. While mechanical windscreens can provide significant signal-to-noise ratio gains, additional benefits can be obtained by employing adaptive filters. In this presentation, a kernel-based adaptive filter based on the Matérn Covariance Function is demonstrated to improve the probability of detection of transient infrasound signals as well as improve signal estimation errors without increasing false alarm rates in the presence of wind noise. The choice of the Matérn Covaraiance Function to represent the wind noise process is motivated by its roots in fractional-order stochastic differential equations. Because the wind noise at infrasound frequencies can be influenced by shear-turbulence interaction and turbulence-turbulence interaction more than one kernel is required to obtain optimal performance for the filter. Results are presented under a range wind noise conditions.

2:00

3pPA4. Calculated wind noise for semi-porous fabric domes. JohnPaul R. Abbott (NCPA, Univ. of MS, 100 Bureau Dr. Stop 8361, Gaithersburg, MD 20899-8361, johnpaul.abbott@gmail.com), Richard Raspet (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of MS, Oxford, MS), John Noble, W. C. Kirkpatrick Alberts, and Sandra Collier (US Army Res. Lab., Adelphi, MD)

A simple calculation of the wind noise measured at the center of three 2 m diameter semi-porous fabric domes is developed. The calculation provides a model of the measured wind noise and is based on the model for a large porous wind fence enclosure. The model combines the wind noise contributions from (a) the turbulence-turbulence and turbulence-shear interactions inside the domes, (b) the turbulence interactions on the surface of the domes, and (c) the turbulence-shear interactions outside of the domes. Each wind noise contribution is calculated from the appropriate measured turbulence spectra, velocity profiles, correlation lengths, and the mean velocity at the center, surface, and outside of the enclosure. The model is verified by comparisons of the measured wind noise to the calculated estimates of the differing noise contributions and their sum. The calculated estimates indicate that the principle source of low frequency wind noise at the center of the domes is due to the turbulence interactions on the surface of the domes while the turbulence interactions inside the domes and outside of the domes are minimal, and in some cases negligible.

2:15

3pPA5. An acoustic model of NIST's long-wavelength acoustic flowmeter for gas flow in large-diameter pipes. JohnPaul R. Abbott (National Ctr. for Physical Acoust., Univ. of MS, Oxford, MS), Keith A. Gillis (National Inst. of Standards and Technol., 100 Bureau Dr. STOP 8360, Gaithersburg, MD 20899-8360, keith.gillis@nist.gov), Lee J. Gorny (None, Mountain View, CA), and Michael R. Moldover (National Inst. of Standards and Technol., Gaithersburg, MD)

NIST is investigating long-wavelength acoustic flowmeter (LWAF) technology to accurately measure gas flow in large-diameter pipes, such as smokestacks used by coal-burning power plants. To aid in the data analysis and development of the method, we constructed a lumped-element acoustic model of the LWAF based on existing theory for sound propagation in circular ducts, modified to include flow. The model calculates the ratio of the acoustic pressure amplitudes and phase differences between two locations in a partial standing wave downstream of a continuous sound source up to the duct's cut-on frequency. We used the numerical calculations of the reflection coefficient by Munt [J. Sound Vibration **142**, 413–436 (1990)] to model the radiation impedance as a function of flow speed. In the absence of flow, the model was used to calibrate the positions of several microphones in the LWAF. In the presence of flow, the model predicts qualitatively the measured amplitude ratios and phase differences as a function of flow rate. Quantitative comparison is limited by the uncertainty of the radiation impedance and its flow dependence. This limitation prompts us to investigate ways to either measure the radiation impedance or eliminate it by using multiple coherent sound sources.

2:30

3pPA6. Feasibility of a long-wavelength acoustic flowmeter for measuring smokestack emissions. John Paul R. Abbott (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., Rm. 1044, Oxford, MS 38677, johnpaul.abbott@gmail.com), Keith A. Gillis (National Inst. of Standards and Technol., Gaithersburg, MD), Lee J. Gorny (None, Mountain View, CA), and Michael R. Moldover (National Inst. of Standards and Technol., Gaithersburg, MD)

Conventional gas flow measurements in large ducts, such as power plant smokestacks, have uncertainties of 5–20%. As part of its Greenhouse Gas and Climate Science Measurements Program, the National Institute of Standards and Technology (NIST) is testing long-wavelength acoustic flowmeters (LWAFs) as an alternative method to reduce this uncertainty. A LWAF uses the Doppler Effect to determine the speed of sound c and the average flow speed V. Theory predicts that, for plane waves in a duct, corrections due to flow irregularities, such as swirl and turbulence, are proportional to $(V/c)^2$ £ 0.01. To investigate the feasibility of using an LWAF in a smokestack, we constructed a 1:100 scale model (10 cm diameter) test facility that generated flows up to 25 m/s using ambient air. The model LWAF simultaneously determined the speed of sound in air with a standard uncertainty of 0.01%, relative to NIST's standard reference database, and measured the average flow velocity with a standard uncertainty of $\pm 1\%$ relative to a NIST-calibrated flow standard upstream from the LWAF. Similar results were obtained when the flows were highly distorted by elbows and obstructions, or when water was sprayed into the air, and for flows through larger diameter model LWAFs.

Session 3pPP

Psychological and Physiological Acoustics: Beyond the Audiogram: Influence of Supra-Threshold Deficits

Agnes C. Leger, Cochair
School of Psychology, University of Manchester, Ellen Wilkinson Building, Oxford Road, Manchester M13 9PL, United Kingdom

Christopher Plack, Cochair
School of Psychological Sciences, University of Manchester, Ellen Wilkinson Building, Oxford Road, Manchester M13 9PL, United Kingdom

Chair's Introduction—1:00

Invited Papers

1:05

3pPP1. Spectrotemporal modulation sensitivity as a predictor of speech intelligibility in noise for hearing-impaired listeners. Joshua G. Bernstein (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 8901 Wisconsin Ave., Bethesda, MD 20889, joshua.g.bernstein.civ@mail.mil)

The audiogram accounts for only a portion of the variance in speech-reception performance in noise for hearing-impaired listeners. The remaining variance is often attributed to a combination of suprathreshold auditory distortion and non-auditory factors such as cognitive processing. This talk describes a series of studies demonstrating that a suprathreshold measure of sensitivity to spectro-temporal modulation (STM) can account for individual differences in speech-reception scores that are not predicted by the audiogram. STM stimuli are spectrally rippled noises with spectral-peak frequencies that shift over time, akin to modulations in a speech signal. The results show that STM sensitivity correlates to speech-reception performance in noise; that the correlation is ascribed mainly to the low-frequency portion of the stimulus (<2 kHz); and that STM sensitivity can account for individual differences in speech-reception thresholds for hearing-impaired listeners properly fit with individualized frequency-dependent gain. Hearing loss has the largest impact on STM sensitivity for low temporal rates and low carrier frequencies, suggesting a reduced ability to use temporal fine-structure information to detect slow-moving spectral peaks. STM detection is a fast, simple test of suprathreshold auditory function that complements the high-frequency audiogram to account for a substantial proportion of individual variability in speech reception in noise.

1:35

3pPP2. Differential effects of noise trauma and diminished endocochlear potential on neural temporal coding of complex sounds: Implications for speech perception. Kenneth S. Henry (Biomedical Eng., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, kenneth_henry@urmc.rochester.edu) and Michael G. Heinz (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

Communication problems due to cochlear hearing loss are pervasive in today's society, even with amplification from digital hearing aids. Threshold elevation may arise from noise-induced trauma to hair cells or, as an alternative mechanism, through age-related depletion of the endocochlear potential (EP; the power supply driving hair-cell function). While both noise trauma and diminished EP may frequently underlie hearing loss in older people, it is unclear whether these pathologies have different effects on neural coding of complex sounds, and ultimately, on speech perception. Here, we describe several recent neurophysiological studies of this question based on Wiener-kernel analyses of chinchilla auditory-nerve fiber responses to Gaussian noise. We find that noise trauma causes pronounced distortions in tonotopic coding of temporal fine structure and slower envelope cues, occurring with as little as 20 dB threshold elevation. Similar changes in neural coding occur with diminished EP, but only when thresholds exceed 60–70 dB SPL. More pronounced changes in temporal coding with noise trauma compared to diminished EP are likely to translate into greater deficits in speech perception (e.g., for noise-induced compared to age-related hearing loss). These results may help explain differences in speech perception abilities across individuals with the same degree of threshold elevation.

2:05

3pPP3. Investigating the role of supra-threshold auditory processing and cognitive abilities in presbyacusis. Christian Fullgrabe (Inst. of Hearing Res., Medical Res. Council, Sci. Rd., Nottingham NG7 2RD, United Kingdom, c.fullgrabe@ihr.mrc.ac.uk)

Anecdotal evidence and experimental investigations indicate that older people experience increased speech-perception difficulties, especially in noisy environments. Since peripheral hearing sensitivity declines with age, lower speech intelligibility can often be explained by a reduction in audibility. However, aided speech-perception in hearing-impaired listeners frequently falls short of the

performance level that would be expected based on the audibility of the speech signal. Given that many of these listeners are older, poor performance may be partly caused by age-related changes in supra-threshold auditory and/or cognitive processes that are not captured by an audiometric assessment. The presentation will discuss experimental evidence obtained from clinically normal-hearing adult listeners showing that auditory temporal processing, cognition, and speech-in-noise perception are indeed linked and, independently of hearing loss, decline across the adult lifespan. These findings highlight the need to take into account audibility-unrelated factors in the prediction and rehabilitation of speech intelligibility across adulthood.

2:35

3pPP4. Evidence that hidden hearing loss does not vary systematically as a function of noise exposure in young adults with normal audiometric hearing. Garreth Prendergast, Hannah Guest, Agnès Léger, Kevin Munro, Karolina Kluk, and Christopher Plack (School of Psychol. Sci., Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, chris.plack@manchester.ac.uk)

Cochlear synaptopathy, or "hidden hearing loss," refers to a loss of synapses between inner hair cells and auditory nerve fibers, and is observed in rodent models as a consequence of noise exposure and/or aging. In humans, cochlear synaptopathy is not thought to be detectable by pure tone audiometry, as thresholds to soft sounds in the rodent models are not permanently elevated. One hundred and forty audiometrically normal participants below the age of 35 and with a range of lifetime noise exposures performed an extensive battery of tests, including electrophysiological measures, psychophysical tests, and speech-in-noise tests. Inter-aural phase discrimination, amplitude modulation detection, and spatial release from masking on a speech task were found to be sensitive to noise exposure; however, these trends are weak and only the phase discrimination task followed the predicted direction (i.e., high noise exposed individuals showing elevated thresholds). None of the electrophysiological measures, including wave I of the ABR, showed a strong relation with noise exposure. The results suggest that either: (i) hidden hearing loss is not prevalent in young normally hearing adults, or (ii) even listeners with comparatively low levels of noise exposure have the disorder and that there is no additional consequence of high levels of exposure.

WEDNESDAY AFTERNOON, 25 MAY 2016 — SALON G, 1:00 P.M. TO 3:00 P.M.

Session 3pSC

Speech Communication: Variation and Gender Effects (Poster Session)

Simone Graetzer, Chair
Communicative Sciences and Disorders, Michigan State University, 1026 Red Cedar Road, Michigan State University, East Lansing, MI 48824

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

Contributed Papers

3pSC1. Effects of low-pass filtering on dialect and gender perception. Robert A. Fox, Ewa Jacewicz, and Zane T. Smith (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

In addition to linguistic (message-related) information, spoken language includes indexical information related to the speaker characteristics (e.g., gender, social status, and regional identity). This study is an extension of Jacewicz *et al.* (2015, *JASA*, **137**, 2417–2418) which explored the nature of acoustic cues signaling indexical information. That study demonstrated that listeners were quite accurate in making decisions regarding the regional dialect and gender of a speaker when responding to short unprocessed phrases from 40 speakers (male and female) from two different dialects spoken in central Ohio and western North Carolina. However, when the signal was low-pass filtered at 400 Hz, sensitivity to both dropped significantly. The current study examined performance on the same phrases when the signal was low-pass filtered at progressively higher cutoff frequencies (500, 700, 900, and 1100 Hz). These stimuli were played to 20 listeners (10 males and 10 females) who identified the sex and dialect of the speaker. As expected, listener sensitivity improved with each wider low-pass filter condition. Performance in the 1100-Hz cutoff condition was very similar to that in the unprocessed condition. Discussion will focus on the nature of the acoustic cues utilized by listeners across the four low-pass filter conditions.

3pSC2. Gender-based variation in the phonetic representation of boundary tones in Persian declarative and interrogative utterances. Marziye Eshghi (Speech, Lang. and Hearing Sci., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Craniofacial Ctr., Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu) and Moharram Eslami (Persian Lang. & Lit., Univ. of Zanjan, Zanjan, Iran)

This study investigated the phonetic representation of boundary tones in Persian declarative and interrogative utterances. Speech samples of the study were selected from Persian speech database, i.e., FARSDAT. These samples were produced by adult Persian speakers ranged in age from 20 to 45 years. PRAAT software was used for measuring the variation of F0 per millisecond at boundary tones. For investigating the phonetic representation

of declarative sentences, a group of ten men and ten women were selected and each speaker produced 10 declarative utterances. Another group of ten men and ten women were selected for the investigation of phonetic representation of boundary tones in interrogative utterances. Each participant produced a yes-no question and a wh-question. Statistical analysis revealed significant difference ($p < 0.05$) between men and women regarding the variation of F0 at the boundary tones of declarative sentences and wh-questions. F0 was observed to fall faster at the boundary tones of declarative sentences and wh-questions produced by women. However, there was no statistically significant difference between men and women for variation of F0 at the boundary tones of yes-no questions. Results could be applied in speech recognition, text-to-speech synthesis, and other areas of speech processing.

3pSC3. The acoustics of charismatic voices in Korean political speech: A cross-gender study. Nari Rhee and Rosario Signorello (Head and Neck Surgery, Univ. of California, Los Angeles, 31-20 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095-1794, n.rhee94@gmail.com)

A previous study (JASA **136**(4), 2295) investigated cross-cultural (Italian, French, and Brazilian) acoustic voice profiles in political speech to discern innate versus learned factors underlying the acoustics of charismatic voices. To further extend the study to other cultures and to both genders the present study investigates how Korean leaders manipulate their voices to exhibit charisma. The voice acoustic profiles of South Korean female (Geunhye Park) and male (Jae-in Moon) politicians were analyzed. Results support previous findings showing a significant cross-culture and cross-gender similarity in communication context-specific (political campaign monologue, political peers-addressed speech, and non-political interview) use of voice F0 and SPL ranges. Data also confirm significant cross-gender differences in speakers' F0 average frequencies (higher for the female speaker) within communication contexts. Finally, results confirm speaker-specific manipulations of F0 and SPL over time. The cross-cultural similarities in charismatic voice showed by the data support our hypothesis of several ongoing cross-cultural/gender processes of integration among some innate and cultural/linguistic vocal habits to convey leadership status: leaders estimate audience's biological and social traits, manipulate vocal ranges of F0 and SPL accordingly, while maintaining women's and men's particular acoustic average features of F0 and SPL significantly different. [Work supported by NIH grant DC01797.]

3pSC4. Region, gender, and within-category variation in American English voiced stops. Abigail H. Elston, Katherine Blake, Kelly Berkson (Linguist, Indiana Univ., 1021 E. 3rd St., Memorial Hall 322 E, Bloomington, IN 47405, abelston@indiana.edu), Wendy Herd, Joy Cariño (English, MS State Univ., Starkville, MS), Max Nelson, Alyssa Strickler (Linguist, Indiana Univ., Bloomington, IN), and Devan Torrence (English, MS State Univ., Starkville, MS)

The two-way voicing contrast in American English stops—particularly in initial position—is often described as a long-lag (e.g., long positive VOT for /p/) versus short-lag (e.g., short positive VOT for /b/) distinction, with less frequent instances of lead voicing (e.g., negative VOT for /b/) attributed to individual variation. Systematic within-category gender and region differences have been reported, however, with more closure voicing found for male than for female speakers (Ryalls, Zipprer, and Baldauff, 1997), and more fully voiced closures for /b/ in female speakers from North Carolina than those from Wisconsin (Jacewicz, Fox, and Lyle, 2009). With this in mind, we investigate the interaction of gender and region in the prevoicing of word-initial voiced stops by comparing the VOTs of male and female speakers from Indiana and Mississippi. Participants were recorded reading three repetitions of a pseudo-randomized list of words including *bot*, *dot*, and *got*. Regional—but no gender—differences were found: speakers from Mississippi produced stops with negative VOT more often (~35% of the time) than speakers from Indiana (~15%), suggesting that southern varieties of English are indeed more heavily prevoiced than other varieties of English.

3pSC5. The voice acoustics of the 2016 United States presidential election candidates: A cross-gender study. Rosario Signorello and Nari Rhee (Head and Neck Surgery, Univ. of California, Los Angeles, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, rsignorello@ucla.edu)

The present study investigates the acoustic voice profiles (vocal fundamental frequency and intensity) of two female (Hillary Clinton and Carly Fiorina) and two male (Bernie Sanders and Donald Trump) candidates to the 2016 United States presidential election. The criteria used for choosing the speakers were based on: (a) an online survey we conducted in 2015; (b) the prediction markets' statistics; (c) the number of endorsements from party elites; (d) the Iowa and the New Hampshire polls; (e) the amount of money raised by the candidate; and (f) the candidates' gender from both Democratic and Republican parties. Voice stimuli for each of these speakers were collected for three different communication contexts: a political campaign monologue; a political peers-addressed speech; and non-political interview. Results confirm and expand previous findings [JASA **136**(4), 2295] showing cross-cultural and gender similarities in voice acoustics supporting the hypothesis of several ongoing cross-cultural/gender processes of integration among some innate and cultural/linguistic vocal habits to convey leadership status: leaders estimate audience's biological and social traits, manipulate vocal ranges of F0 and SPL accordingly, while maintaining women's and men's particular acoustic average features of F0 and SPL significantly different. [Supported by NIH grant DC01797.]

3pSC6. Stylistic variation in children's vowel production. Ewa Jacewicz, Robert A. Fox, and Jill M. Deatherage (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

The monophthongization of the diphthong [ai] to [a] is possibly the most stereotyped feature of Southern American English. This feature can be observed to varying degrees across the American South. Although monophthongization represents an important social symbol in the South, its pervasiveness has begun to recede in recent generations. This sound change has been brought on by multiple factors including population mobility, education, and urbanization, which have promoted acceptance of a more standardized variety of American English, including diphthongal realizations of [ai]. The current study examines the change in progress toward [ai] in 20 children aged 8–10 from western North Carolina in three different speaking styles: citation-form words, read sentences, and informal, unscripted talks. In addition to acoustic analysis, stylistic variation was evaluated by 20 central Ohio listeners (10 males and 10 females). Participants were asked to listen to individual words and to rate the degree of perceived monophthongization on a seven-point scale. Results showed an increase in perceived monophthongization from citation forms to informal talks. Compared to girls, boys had a higher occurrence of the monophthongal variants and lower occurrence of the full diphthongs across all production types. Discussion will focus on the advancement of the sound change.

3pSC7. Canadian raising in Fort Wayne, Indiana. Alyssa Strickler, Kelly Berkson, and Stuart Davis (Linguist, Indiana Univ., 1021 E. Third St., Memorial Hall 322E, Bloomington, IN 47405, arstrick@indiana.edu)

The phenomenon often referred to as Canadian Raising, wherein speakers raise the diphthongs /aɪ/ and /aʊ/ to [ʌɪ] and [ʌʊ], respectively, when preceding voiceless sounds, is attested not only in Canada but also in places in the Northern periphery of the U.S. (e.g., Ann Arbor and the Upper Peninsula, in Michigan). Raising has also been documented in locations distant from the border, such as Philadelphia. The current study examines the spread of raising farther south from Michigan into northeast Indiana, specifically Fort Wayne. Acoustic analysis of preliminary data from Fort Wayne area speakers was conducted to determine whether raising indeed occurs and to explore the effect of contextual factors such as speaker sex, age, and prosodic structure. The results confirm that while this raising does in fact happen around Fort Wayne, there is an age effect: younger speakers (<25 yr) show raising across the board in the expected environments but older speakers (>60 yr) do not. This suggests that vowel raising is a relatively new characteristic of Fort Wayne English.

3pSC8. The effect of structural context on phonetic accommodation. Jonathan T. Manker (Dept. of Linguist, Univ. of California, Berkeley, Berkeley, CA 94720, jtmanker@berkeley.edu)

Phonetic accommodation is a linguistic phenomenon whereby phonetic characteristics of one's speech are influenced by perceiving the speech of others. The last decade has seen a multitude of studies which have induced the phenomenon in a lab setting, considering the social environments necessary for facilitating or inhibiting it. No studies, however, have investigated how the perception and production of speech in different structural contexts affects phonetic accommodation. This presentation analyzes the results of an experiment that compared the frequency of phonetic accommodation in three speech settings. In condition one, subjects read aloud and shadowed entire sentences which contained artificially lengthened VOT. In condition two, subjects read short phrases and shadowed sentences, and in condition three, both read and shadowed phrases. Our results showed that speakers accommodated lengthened VOT but often also showed modified burst intensities, suggesting imprecise accommodation. More accurate VOT accommodation was observed in conditions 2 and 3 (9/12 subjects for both) compared to condition 1 (5/12 subjects). "Inaccurate" burst accommodation was more common in condition 1 (8/12) and 2 (8/12) compared to condition 3 (4/12). The results suggest that higher levels of syntactic and semantic processing (both in production and perception) may impede processing of phonetic details.

3pSC9. Dissociating contributions of talker gender and acoustic variability for spectral contrast effects in vowel categorization. Ashley Assgari, Asim Mohiuddin (Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, ashley.assgari@louisville.edu), Rachel Theodore (Univ. of Connecticut, Storrs, CT), and Christian Stilp (Univ. of Louisville, Louisville, KY)

Speech categorization is influenced by spectral contrast effects (SCEs), the perceptual magnification of spectral differences between successive sounds. Through SCEs, preceding acoustic contexts can bias categorization of following sounds away from reliable spectral properties. Recent findings (Assgari & Stilp, 2015 JASA) show that SCEs in vowel categorization can be modulated by talker characteristics: a clear SCE when the preceding context was 200 sentences spoken by a single talker was diminished when the context featured 200 talkers. This result was attributed to variability in mean pitch of the preceding sentences. However, neither mean sentence pitch nor talker gender was explicitly controlled, which challenges identification of the locus of the talker effect. The current study examined whether gender and pitch variability yield separable contributions to SCEs. Listeners heard precursor sentences then categorized a target vowel from an /ɪ/ to /ɛ/ continuum. Sentences were processed to add a modest low-F1 or high-F1 spectral peak to encourage "eh" or "ih" responses, respectively. Critically, talker gender was crossed with pitch variability in these sentences. Results will isolate the property responsible for attenuating SCEs if gender and pitch variability have dissociable effects. Failure to observe this pattern suggests similar effects of these properties on SCEs.

3pSC10. Gender and rate effects on speech intelligibility. Eric M. Johnson and Sarah H. Ferguson (Dept. of Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, eric.martin.johnson@utah.edu)

Older adults seeking hearing help often complain of particular difficulty understanding female voices. This contrasts with studies using young listeners with normal hearing in which female talkers have been found to be generally more intelligible than male talkers (e.g., Bradlow *et al.*, 1996). Could some factor in addition to talker gender be causing older adults to have increased difficulty understanding female voices? Speech that has been time-compressed has been shown to be less intelligible than unprocessed speech (e.g., Gordon-Salant and Friedman, 2011), but few data exist to show whether an increased presentation rate causes an equal loss of intelligibility for male and female talkers. The present study will explore whether an increased playback rate has a greater negative effect on the intelligibility of speech produced by female versus male talkers. Subjects will listen to sentences produced by two female and two male speakers from the Utah Speaking Style Corpus presented at either their original rate or at an increased rate (1.5 times faster) and type out what they heard. The resulting data will show whether, for young normal-hearing listeners, an increased rate of speech affects the intelligibility of female talkers more than it affects the intelligibility of male talkers.

3pSC11. Time dose and fundamental frequency of male and female speakers in different style and reverberation times. Pasquale Bottalico, Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48910, pb@msu.edu), and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Speakers adjust their vocal production when communicating in different room acoustics, when instructed to speak at different volumes and when experiencing vocal fatigue. The present paper reports on the effects of speech style, reverberation time, and external auditory feedback on time dose and fundamental frequency. Ten male and ten female subjects were recorded while reading a text in normal and loud styles, in three rooms—anechoic, semi-reverberant, and reverberant—and with and without acrylic glass panels at 0.5 m from the mouth, which increased external auditory feedback. Longer time doses were accumulated in more reverberant rooms, especially when loud voice was used. Higher fundamental frequency was measured in less reverberant rooms. Subjects increased their fundamental frequency in the loud speech style versus the normal style, but this effect was weaker when the level of external auditory feedback was high. A larger fundamental frequency range was detected for females than males and for the loud style than the normal style. These results contribute to understanding how the effect of room acoustics on speech changes as a function of speaking time.

3pSC12. Perception of voice gender in cochlear implant simulations of children's speech. Daniel R. Guest, Michelle R. Kapolowicz, Shaikat Hossain, Vahid Montazeri, and Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, GR41 The University of Texas at Dallas, Box 830688, Richardson, TX 75083, daniel.guest@utdallas.edu)

Previous studies [Assmann *et al.*, *J. Acoust. Soc. Am.* **135**, 2424 (2014) and Assmann *et al.*, *J. Acoust. Soc. Am.* **138**, 1811 (2015)] investigated normal-hearing listeners' ability to discriminate gender and age in children's speech. The speech stimuli were /hVd/ syllables produced by 140 speakers, ages 5 through 18, and processed using the STRAIGHT vocoder to simulate a change in speaker gender. Experimental conditions involved swapping the fundamental frequency contour (F0) and/or the formant frequencies (FF) to the opposite-sex average within each age group. The present study extended these previous experiments by presenting the stimuli to normal-hearing listeners through a cochlear implant simulation implemented as a sine wave vocoder. Supporting findings reported at previous meetings, swapping F0 has a larger effect on perceived gender than swapping FF for older voices, and single-parameter changes (swapping either F0 or FF), had relatively small effects on older male voices but pronounced effects on older female voices. Overall, the present findings indicate that gender recognition is more difficult in a cochlear implant simulation. Furthermore, gender recognition in a cochlear implant simulation is particularly difficult with ambiguous voices, such as those of young children.

Session 3pUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Sediment Characterization Using Direct and Inverse Techniques II

David P. Knobles, Cochair
None, KSA LLC, PO Box 27200, Austin, TX 78755

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Contributed Papers

1:30

3pUW1. Field measurements of interface waves and geoacoustic characterization of near surface sediment and soil. Jennifer Giard, Gopu R. Potty, James Miller, Christopher Baxter (Ocean Eng. Dept., Univ. of Rhode Island, 215 S Ferry Rd., Narragansett, RI 02882, jennifer_giard@my.uri.edu), and Aaron Bradshaw (Civil and Environ. Eng., Univ. of Rhode Island, Narragansett, RI)

There is a need for rapid and nondestructive sensing of near-surface shear properties of the ground and seafloor. Our approach is to measure interface wave dispersion and invert these measurements to extract a shear wave speed profile. Field measurements of interface waves using geophones and accelerometers will be presented. A laser Doppler vibrometer (LDV) will also be incorporated in the suite of sensors for the measurement of wave properties. Data from these sensors will be compared to understand the coupling of the sensors into the soil or sediment and its effect on the measurements. Geoacoustic and geotechnical properties will be estimated using the interface wave data. The uncertainty and resolution of the estimates in different sediment/soil types will be explored. The Interface Wave Testing Facility at the University of Rhode Island will also be used to understand the surface wave propagation characteristics in different type of sediments/soils. [Work supported by Army Research Office and the Office of Naval Research.]

1:45

3pUW2. Development of a system for *in situ* measurements of geoacoustic properties during sediment coring. Megan S. Ballard, Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Sediment cores provide valuable insight on the physical properties of the seabed, and laboratory measurements of sediment wave speed from cores are often considered "ground truth." However, sound-speed estimates obtained from cores can be inaccurate due to changes in pressure, temperature, and mechanical properties of the sediment caused by removal of the core from the seabed and its subsequent transport to the laboratory. To begin to address this deficiency, we report on the development of a system for obtaining *in situ* measurements of geoacoustic properties. The system mounts on the nose of a coring barrel to obtain an *in situ* record of compressional and shear wave speed and attenuation as the core penetrates the seabed. The depth of the *in situ* record is limited only by the penetration of the core. The compressional wave measurements are obtained with rod-mounted piezoelectric cylinders, and the shear wave measurements are obtained with bender elements mounted in flat blades. For both the compressional and shear wave measurements, wave speed and attenuation are estimated from differential measurements made with two receivers. [Work supported by ONR.]

2:00

3pUW3. Geoacoustic inversion based on particle velocity. David R. Dall'Osto and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th, Seattle, WA 98115, dallosto@uw.edu)

An approach to geoacoustic inversion using a non-dimensional vector quantity, known as circularity, is discussed. This quantity corresponds to the difference in phase and magnitude of the vertical and horizontal components of particle velocity, and is also the normalized curl of the active intensity vector. As such, it is highly dependent on the phase, amplitude, and arrival angle of interfering multipath arrivals, with value as a function of source depth that depends on both frequency and bottom type. We first show results from the Targets and Reverberation Experiment (TREX) which took place off Panama City, where frequencies in the range of 1–4 kHz are used to study sandy sediments [Dall'Osto *et al*., J. Acoust. Soc. Am. **139**, 311–319, 2016]. The approach is next applied hypothetically at lower frequencies for which fewer modes are supported in the waveguide, and to different seabed conditions to demonstrate how a range of sediments from sand to mud are distinguished.

2:15

3pUW4. Sediment characterization from normal incidence bottom loss at the GLISTEN sea test. Marcia J. Isakson, James Piper, and Roger Banks (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

The GLider Sensors and payloads for Tactical characterization of the ENvironment (GLISTEN) sea test was conducted off the west coast of Italy in August of 2015. Led by the Centre for Maritime Research and Experimentation (CMRE), the effort included seven other international organizations including the Applied Research Laboratories from The University of Texas at Austin (ARL:UT). The mission of the sea test was to improve and understand long-range acoustic methods of environmental characterization in shallow waters. As part of the test, a team from ARL:UT deployed a remotely operated vehicle (ROV) which collected normal incidence bottom loss data from 6 to 20 kHz along with video and laser line profiling data. These data were taken along the propagation paths of the CMRE low-frequency transmission loss experiment. In this presentation, results will be shown from the bottom loss measurements for a variety of sediments that included both layering and entrained gas pockets. The acoustic data will be compared with models that include such frequency dependent effects as interface roughness and volume scattering. [Work supported by ONR, Ocean Acoustics.]

2:30

3pUW5. Spatial variability of sediment roughness and its effect on reverberation. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

As part of the environmental characterization supporting the Target and Reverberation Experiment in 2013 (TREX13), the Seafloor Laser-line Scanner (SLS) was deployed to measure sediment roughness at locations throughout the experiment site. The SLS uses structured light to create a two-dimensional digital elevation map of the seafloor over a 0.3 $\times$ 3.5 m area. This map is processed to determine the seafloor roughness power spectrum which is used to model backscatter from the seafloor. While a relatively shallow site for TREX13 was chosen to limit the area within which environmental characterization was required, there was significant variability in the sediment roughness which could not be sufficiently measured in the time available. To overcome this difficulty, data from a 400 kHz multibeam backscatter survey collected by de Moustier and Kraft have been combined with the SLS measurements to infer the roughness throughout the site. Although the reverberation modeling focuses on scattering wavenumbers that affect the 2–4 kHz acoustic measurements, the 400 kHz backscatter strengths are correlated to the spectral strengths at the mid-frequency wavenumbers of interest. The implications of these results for reverberation modeling as well as the limits of this approach will be discussed. [Work supported by ONR.]

Plenary Session and Awards Ceremony

Christy K. Holland, Chair
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

Ian C. Bruce – For contributions to models of auditory-nerve fibers

Micheal L. Dent – For contributions to spatial hearing in animals

Christine Erbe – For contributions to the effects of anthropogenic noise on marine mammals

Kent L. Gee – For contributions to jet-noise and nonlinear propagation

Karen S. Helfer – For contributions to speech perception in aging

Eric J. Hunter – For contributions to laryngeal function for voice production

G. Christopher Stecker – For multidisciplinary contributions to binaural hearing

Introduction of Award Recipients and Presentation of Awards

William and Christine Hartmann Prize in Auditory Neuroscience to Alan R. Palmer

Distinguished Service Citation to Susan B. Blaeser

R. Bruce Lindsay Award to Megan S. Ballard

Helmholtz-Rayleigh Interdisciplinary Silver Medal to Armen Sarvazyan

Gold Medal to Whitlow W. L. Au

Vice President’s Gavel to Lily M. Wang

President’s Tuning Fork to Christy K. Holland

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m. These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Wednesday are as follows:

Committee	**Room**
Biomedical Acoustics	Salon H
Musical Acoustics	Salon B/C
Signal Processing in Acoustics	Salon G

ACOUSTICAL SOCIETY OF AMERICA
DISTINGUISHED SERVICE CITATION

Susan B. Blaeser

2016

The Distinguished Service Citation is awarded to a present or former member of the Society in recognition of outstanding service to the Society.

PREVIOUS RECIPIENTS

Laurence Batchelder	1972
Robert W. Young	1973
Betty H. Goodfriend	1973
Gerald J. Franz	1974
Robert T. Beyer	1978
Henning E. von Gierke	1978
R. Bruce Lindsay	1981
William S. Cramer	1984
Stanley L. Ehrlich	1986
Samuel F. Lybarger	1986
Frederick E. White	1987
Daniel W. Martin	1989
Murray Strasberg	1990
William J. Cavanaugh	1994
John C. Burgess	1996
Alice H. Suter	1997
Elaine Moran	1999
John V. Bouyoucos	2000
F. Avril Brenig	2000
Thomas D. Rossing	2006
Charles E. Schmid	2008
Uwe J. Hansen	2011
Richard Stern	2011
Allan D. Pierce	2015
Paul D. Schomer	2015

CITATION FOR SUSAN B. BLAESER

. . . for dedicated management of the Acoustical Society of America standards program

SALT LAKE CITY, UTAH • 25 MAY 2016

Susan Blaeser has been Standards Manager of the Acoustical Society of America (ASA) for the past fifteen years. During her tenure with ASA, she has been extraordinary in guiding the Standards program, and in representing ASA at national and international levels. She provides excellent leadership and guidance to every activity in which she takes part, not just Standards. The breadth and depth of her skills, actions, and service to ASA are best conveyed with a few examples.

Susan's career with ASA began at the close of the ballot on the classroom acoustics Standard which was contentious, to say the least. The appeals process took years and many hours of work by a select team of ASA Standards leaders including Richard Stern, then President of ASA. Susan got up to speed on day one of her tenure here at ASA, and on day two she was contributing fully to the group with suggestions of concepts, tactics, and strategies in addition to keeping the process moving along and managing all the details.

A little over eight years ago, when the American National Standards Institute (ANSI) wanted to impose a large increase in their fee for royalties related to sales of Nationally Adopted International Standards, Susan rallied many of the similarly affected Standards Developing Organizations (SDOs), and together, they succeeded in mitigating the increase. Currently, ANSI is again looking to increase the fees its members pay, but this time they took many steps to engage with the participants to plan the new funding structure and included ASA in this consultation alongside the larger Standards Developers. Susan has created a space for the much littler ASA among the giants because of her ideas, leadership, and service.

Several years ago, Susan worked diligently with several ASA Technical Committees to create a new national standards subcommittee on Animal Bioacoustics, S3/SC 1. This involved outreach to identify stakeholders both inside and outside the ASA, who would form the committee and participate as voting members. Shortly thereafter, in response to requests from the underwater technical community, Susan along with the Standards Director and several others proposed to the ASA Committee on Standards (ASACOS) the formation of an international subcommittee on underwater acoustics. As soon as ASACOS had discussed and approved this plan, Susan was off and running at full speed to make this plan for a new International Organization for Standards (ISO) subcommittee a reality. The scope of this new subcommittee consists of technical areas that are relevant to several ASA Technical Committees. This committee is now ISO/TC 43/SC 3, chaired by George Frisk. Again, Susan invested in outreach to stakeholders from countries around the world to help them engage their national standards bodies to take an active role. Currently the ASA (i.e., Susan and her staff) provides the Secretariat for ISO TC43/SC3. This ISO subcommittee began its work by converting an ANSI/ASA Standard into an ISO Standard to deal with the measurement of ship noise. The subcommittee has now expanded to four working groups with additional projects planned. Along the way there was conflict with another ISO committee that hoped to produce its own Standard to measure ship noise without input from the TC 43 Subcommittee 3, which includes underwater acoustics in its scope. Susan waged a relentless campaign for proper recognition of the TC 43 Subcommittee 3. It took nearly three years to get underwater acoustics consolidated in the TC 43 subcommittee and to establish cordial relations between the two groups, but Susan did it. Having the leadership of the underwater acoustics subcommittee provides another position of international leadership for ASA and several of its Technical Committees that would not exist, if not for Susan and her efforts and leadership.

Through the years Susan has been an outspoken representative for ASA on numerous committees and working groups in ANSI and with other SDOs to make sure that ASA's views are incorporated in any actions that are issued in the name of the standards community as a whole, such as the United States Standards Strategy.

While managing the ASA Standards office, serving as the Secretary for three international committees or subcommittees, managing the U.S. input to nine international committees, managing national Standards development, finding new members, and reducing expenses of the ASA Standards program, Susan still always makes time to not only help anyone who asks, but also to offer help where projects seem to be faltering. During the revision of ANSI/ASA S1.1 Acoustical Terminology, the Working Group faced the daunting task of responding to hundreds of comments. It was Susan's energy and exuberance that helped the Chair keep this effort on track and brought it to a successful conclusion.

In another case, a Working Group chair relatively new to Standards was 90% of the way to developing a Standard, but like others, became bogged down in the final comments. Susan completely re-edited the Standard, highlighting issues and inconsistencies. With this in hand, it was a simple matter for her to have a telephone conference with the Chair, and complete the Standard.

These examples are not exceptions, rather they are the norm. Clearly, Susan's capabilities, accomplishments, and service to ASA go beyond our Standards program, ANSI, and standards development in the US, and reach far into the international acoustics community. Susan has been more than just the Standards Manager. She is a full Member of ASA, and one of our key assets. It is the diplomatic and professional manner with which she has handled everything that makes her a remarkable ambassador for ASA to the world. This is her legacy.

PAUL D. SCHOMER

ACOUSTICAL SOCIETY OF AMERICA

R. BRUCE LINDSAY AWARD

Megan S. Ballard

2016

The R. Bruce Lindsay Award (formerly the Biennial Award) is presented in the Spring to a member of the Society who is under 35 years of age on 1 January of the year of the Award and who, during a period of two or more years immediately preceding the award, has been active in the affairs of the Society and has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics, or both. The award was presented biennially until 1986. It is now an annual award.

PREVIOUS RECIPIENTS

Richard H. Bolt	1942
Leo L. Beranek	1944
Vincent Salmon	1946
Isadore Rudnick	1948
J. C. R. Licklider	1950
Osman K. Mawardi	1952
Uno Ingard	1954
Ernest Yeager	1956
Ira J. Hirsh	1956
Bruce P. Bogert	1958
Ira Dyer	1960
Alan Powell	1962
Tony F. W. Embleton	1964
David M. Green	1966
Emmanuel P. Papadakis	1968
Logan E. Hargrove	1970
Robert D. Finch	1972
Lawrence R. Rabiner	1974
Robert E. Apfel	1976
Henry E. Bass	1978
Peter H. Rogers	1980
Ralph N. Baer	1982
Peter N. Mikhalevsky	1984
William E. Cooper	1986
Ilene J. Busch-Vishniac	1987
Gilles A. Daigle	1988
Mark F. Hamilton	1989
Thomas J. Hofler	1990
Yves H. Berthelot	1991
Joseph M. Cuschieri	1991
Anthony A. Atchley	1992
Michael D. Collins	1993
Robert P. Carlyon	1994
Beverly A. Wright	1995
Victor W. Sparrow	1996
D. Keith Wilson	1997
Robert L. Clark	1998
Paul E. Barbone	1999
Robin O. Cleveland	2000
Andrew J. Oxenham	2001
James J. Finneran	2002
Thomas J. Royston	2002
Dani Byrd	2003
Michael R. Bailey	2004
Lily M. Wang	2005
Purnima Ratilal	2006
Dorian S. Houser	2007
Tyrone M. Porter	2008
Kelly J. Benoit-Bird	2009
Kent L. Gee	2010
Karim G. Sabra	2011
Constantin-C. Coussios	2012
Eleanor P. J. Stride	2013
Matthew J. Goupell	2014
Matthew W. Urban	2015

CITATION FOR MEGAN S. BALLARD

. . . for contributions to underwater acoustic propagation modeling and inversion techniques in acoustical oceanography

SALT LAKE CITY, UTAH • 25 MAY 2016

Megan S. Ballard was born in Grand Rapids, Michigan and earned her B.S. degree (*magna cum laude*) in Ocean Engineering in 2005 from Florida Atlantic University, under the mentorship of George Frisk. After serving as an undergraduate research assistant in 2002, she worked as a summer intern, honing her practical skills, at the Harbor Branch Oceanographic Institution, Ft. Pierce, Florida in 2003, at the U.S. Naval Sea Systems Command in Washington, DC in 2004, and at the Lockheed Martin Corp., Riviera Beach, Florida in 2005. While at Florida Atlantic, Megan continued a lifelong goal to achieve excellence, earning a place on the Dean's List–every semester, winning the Florida Bright Future scholarship (2000–2005), the Society of Naval Architects and Marine Engineers Scholarship (2004), Scholar for the College of Engineering (2004), the Charles Stephan Award, and the Kenneth R. Williams Leadership Award (2005). No wonder that Professor Frisk calls her "his best undergraduate student, ever."

Megan began graduate studies at the Pennsylvania State University (PSU) in August 2009, conducting her dissertation under the supervision of Kyle Becker on the topic of geoacoustic inversion, which yields geoacoustic parameters of the sea floor from inversion of acoustic propagation measurements. While at PSU, she garnered the National Defense Industrial Association Fellowship Award for Undersea Systems (2006), a National Defense Science and Engineering Fellowship (2006–2009), three Acoustical Society of America (ASA) Best Student Paper Awards (2007 and 2008), as well as the Simowitz Citation of the PSU Acoustics Program. No wonder that Professor Becker calls her "his best graduate student, so far." She earned her Ph.D. in the PSU Acoustics Program in 2009.

Her dissertation work led to four archival publications in the *Journal of the Acoustical Society of America* (JASA) and she has continued to develop and extend that work today. Since the dissertation, Megan extended her inversion technique for use in three-dimensional water column sound speed estimation, partly inspired by the elevated interest in ocean circulation modeling associated with the Deep Water Horizon disaster. That work led to another JASA paper.

After her graduate work, Megan joined the Applied Research Laboratories of the University of Texas at Austin (ARL:UT Austin) and was a Postdoctoral Fellow, 2010–2011, working in ocean propagation modeling and inversion. She became an ARL:UT Austin staff member in 2012 and continues her modeling work as well as new work in sonar engineering and in sediment acoustics. Her three dimensional inversion expertise led to her first M.S. thesis supervision of Christopher Bender at UT Austin in 2012.

Megan's sediment acoustics work involves the propagation of sound and shear waves in mud and silty clay sediments. Here, she has been active in both theory and experiment. Her theoretical work has involved the fascinating "House of Cards" concept, advanced by the late William Carey and by Alan Pierce and William Siegmann, whereby the thin clay platelets are held together by electric charges at their ends and centers. As a result of her work, we now know that shear waves exist in soft mud bottoms – an important result, since they can extract energy from propagating waves in the overlying seawater. She has made applications of the House of Cards theoretical model, and has participated with others in laboratory and field experiments on compressional and shear wave velocity and attenuation in clays and mud sediments.

Her laboratory experiments have included electron microscope measurements of the properties of kaolinite platelets, including size distributions and chemical properties. She has incorporated these measurements to evaluate House of Cards models, and was lead author of a JASA paper on this topic.

Field measurements were conducted by Megan and her colleagues in Currituck Sound, behind the North Carolina barrier island, at the U.S. Army Corps of Engineers Field Research Facility at Duck, North Carolina. This work has produced a JASA manuscript and continues today.

Part of Megan's success is due to fearlessness in attempting new things and a formidable determination to achieve excellence in all aspects of her work. She eagerly addresses difficult problems that are not within her sphere of past experience, as illustrated by her evolution from numerical modeler to a laboratory and field experimentalist, and her undertaking of sonar engineering tasks.

Megan's achievements and leadership roles in the ASA are consistent, commendable, and remarkable for a young professional of her age. She started at the bottom and worked her way up. Just a few highlights: Secretary of the Central Pennsylvania ASA Chapter (2006), Student Council member (2007–2009), Acoustical Oceanography Technical Committee member (2009–present), Women in Acoustics Committee member (2009–present), Associate Editor of *Proceedings of Meetings on Acoustics* (POMA, 2014–present), and finally she has just been elected Chair of the Underwater Acoustics Technical Committee (2015–present). She has volunteered to chair and co-chair numerous special sessions at ASA meetings and has made significant efforts to stimulate professional interaction on Society matters, of both scientific and organizational nature. She has attended and presented papers at Society meetings, as well as meetings of other societies in both the US and abroad.

Megan is at the forefront of her chosen research arena in geoacoustic inversion and sediment acoustics, having already published 15 archival papers and presented, authored, or co-authored a like number of meeting and conference proceedings. Megan presently leads a small group at ARL:UT Austin, where she has supervised students, including a Master's degree candidate as well as several summer student scholars. She is married to Jeffrey Ballard and they have two young daughters. She is also an early riser and an avid, long-distance runner.

We are pleased to congratulate Megan Ballard, on behalf of her ASA colleagues, her co-workers, former professors, and students, on her selection as the 2016 recipient of the Acoustical Society of America's R. Bruce Lindsay Award.

THOMAS MUIR
PRESTON WILSON

ACOUSTICAL SOCIETY OF AMERICA
HELMHOLTZ-RAYLEIGH INTERDISCIPLINARY SILVER MEDAL
in

Physical Acoustics, Biomedical Acoustics, and Engineering Acoustics

Armen Sarvazyan

2016

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler	1997	Mathias Fink	2006
David E. Weston	1998	Edwin L. Carstensen	2007
Jens P. Blauert	1999	James V. Candy	2008
Lawrence A. Crum	2000	Ronald A. Roy	2010
William M. Hartmann	2001	James E. Barger	2011
Arthur B. Baggeroer	2002	Timothy J. Leighton	2013
David Lubman	2004	Mark F. Hamilton	2014
Gilles A. Daigle	2005	Henry Cox	2015

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993

CITATION FOR ARMEN SARVAZYAN

. . . for contributions to ultrasound imaging and its applications

SALT LAKE CITY, UTAH • 25 MAY 2016

Armen Sarvazyan has lived an interesting life. Born in Yerevan, Armenia, which was part of the USSR, and attended the prestigious Moscow State University where he earned a B.S. in Physics in 1962 and an M.S. in Biophysics in 1964. He was also an active visual artist–sculpting, and drawing. As a graduate student and researcher at the Pushchino Research Center of the USSR Academy of Sciences, Armen earned his Ph.D. in 1969 but also founded and was first president of the Pushchino Arts and Crafts Center “Koryaga.” His artistic and scientific efforts embraced the same purpose: to understand and honor nature, while finding creative solutions to human problems. As such, Armen eventually became one of the world’s foremost inventors, with over 100 patents and 200 peer-reviewed scientific publications, and the “godfather” of the field of ultrasound elastography.

Armen rose quickly from junior scientist, to senior scientist, to head of the Biophysical Acoustics Laboratory in the Institute of Theoretical and Experimental Biophysics, Russian Academy of Sciences in Pushchino, Russia. There he supervised the work of over 20 Ph.D. students and taught courses in physics, acoustics, and biomedical systems. He even served a stint on a Soviet commission to investigate paranormal phenomena, which yielded many interesting experiences including meeting people with astounding mental capabilities and encountering unusual physical phenomena such as ball lightning. Armen notes that “everything interesting that we investigated was not repeatable; everything repeatable was not interesting.”

At that time, travel outside of the USSR by Soviet scientists was nearly impossible. Instead, scientists would come to visit Pushchino; among the US acousticians hosted by Armen were Floyd Dunn, Frank Fry, Wes Nyborg, Kit Hill, Mack Breazeale, Bill O'Brien, Roy Williams, and Christy Holland, as well as leading biomedical acoustics researchers from Eastern Europe. These visitors inspired Armen’s investigations of, for example, nonlinear acoustics, which led to papers on the molecular mechanisms of acoustic nonlinearity and techniques to measure them precisely. Similarly, Armen first became acquainted with the acoustic radiation force as a possible ultrasound bioeffect in discussions with Leonid Gavrilov and Frank Fry.

Researchers in ultrasound were concerned with tissue characterization, that is, determining experimental characteristics of human tissue that could aid in diagnosis and therapy. Armen Sarvazyan was among the first to realize that there was significant structural information in the shear properties of soft tissue, and that these properties could be sensitively probed using low-frequency shear waves. At Pushchino in the 1970s and 1980s, Armen developed low-frequency acoustic methods and analytic tools to study the shear properties of soft tissue and correlate shear properties with disease characteristics of many tissue types. This work led directly to the later development of elasticity imaging, or elastography, a multi-billion-dollar branch of biomedical ultrasound that provides detailed diagnostic images from shear wave interactions, for instance, in cancer tumor detection. Virtually every clinical ultrasound imaging machine now includes shear wave imaging capability.

Early shear wave transducers used a piezoelectric crystal that moved parallel to its contact face. Devices and analytical tools pioneered by Armen and colleagues helped to solve the inverse problem of determining the characteristics of deeper objects: the “princess and the pea” problem. But in searching for better ways to move tissue, Armen Sarvzyan was among the first to use acoustic radiation force to generate the low-frequency shear waves. Many researchers thought this would not work because of the small magnitude of the radiation force in tissue, but being an inventor and engineer, Armen developed a method described by his seminal patent in 1997, “Method and device for shear wave elasticity imaging.” His 1998 archival paper describing this method, co-authored by colleagues at the University of Michigan, is a thousand-citation classic that opened the field to clinical practice and spawned several companies.

Emigrating to the United States, Armen Sarvazyan first took a position as research professor and head of the Bioacoustics Laboratory in the Chemistry Department at Rutgers University. In 1994, he founded Artann Laboratories, Inc., for which he is still the chief scientific officer. The enormous engineering accomplishments of Artann Labs include over 60 US patents and development of devices to diagnose prostate and breast cancer. At Artann Labs, considerable advances have been made in nonlinear focusing by time-reversal acoustics, ultrasound-assisted drug delivery to brain tumors, assessment of body hydration status, and monitoring of colonoscopy force patterns, among many areas. Other companies, often led by Armen's former students or collaborators, have grown up from the ultrasonic methods and devices that he developed and patented. For example, a company in Israel is commercializing a device for monitoring and analyzing liquid food products such as milk, juices, and other beverages. An Irish company is commercializing a device for monitoring biomolecular structural changes in the pharmaceutical, biotech, polymer, and petrochemical industries. A different ultrasonic device is installed in hundreds of Russian hospitals for determining the molecular composition of blood serum and whole blood that does not require costly reagents and takes minutes instead of hours to complete analysis. US companies have developed technologies based on the use of cylindrical ultrasonic resonators and other devices for monitoring various processes of biochemical and/or pharmaceutical importance such as drug-DNA interactions, thermodynamics of conformational transitions of biopolymers, rapid detection of food borne pathogenic bacteria, and other highly sensitive immunochemical sensors. Armen's engineering accomplishments extend to such diverse fields as the artificial insemination of pigs, manufacturing of golf balls and sausages, land mine detection, and development of non-lethal weapons.

Armen has maintained an ardent connection with the Acoustical Society of America (ASA) over the years, presenting more than 60 talks at ASA meetings with about one-third invited. A Fellow of the ASA, he can be found at most meetings sharing ideas and proposing innovative solutions to difficult problems.

Armen's artistic production never stopped, either, and has been recognized in international art festivals and exhibited in galleries in New York City. Working in media as diverse as wood, stone, concrete, paint, and found objects, Armen has examined the relationships of classic form with a creative (and often humorous) spin. Some of it can be viewed at www.armensarvazyan.net. Whether investigating molecular interactions using carefully designed resonators or creating clever art sculptures, Armen Sarvazyan has helped elucidate nature's beauty and turn it to good purpose for humanity's benefit.

We are pleased to congratulate Armen Sarvazyan for being awarded the ASA Helmholtz-Rayleigh Interdisciplinary Silver Medal in Physical Acoustics, Biomedical Acoustics, and Engineering Acoustics.

E. CARR EVERBACH
LAWRENCE A. CRUM

ACOUSTICAL SOCIETY OF AMERICA

GOLD MEDAL

Whitlow W. L. Au

2016

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

Wallace Waterfall	1954
Floyd A. Firestone	1955
Harvey Fletcher	1957
Edward C. Wente	1959
Georg von Békésy	1961
R. Bruce Lindsay	1963
Hallowell Davis	1965
Vern O. Knudsen	1967
Frederick V. Hunt	1969
Warren P. Mason	1971
Philip M. Morse	1973
Leo L. Beranek	1975
Raymond W. B. Stephens	1977
Richard H. Bolt	1979
Harry F. Olson	1981
Isadore Rudnick	1982
Martin Greenspan	1983
Robert T. Beyer	1984
Laurence Batchelder	1985
James L. Flanagan	1986
Cyril M. Harris	1987
Arthur H. Benade	1988
Richard K. Cook	1988
Lothar W. Cremer	1989
Eugen J. Skudrzyk	1990
Manfred R. Schroeder	1991
Ira J. Hirsh	1992
David T. Blackstock	1993
David M. Green	1994
Kenneth N. Stevens	1995
Ira Dyer	1996
K. Uno Ingard	1997
Floyd Dunn	1998
Henning E. von Gierke	1999
Murray Strasberg	2000
Herman Medwin	2001
Robert E. Apfel	2002
Tony F. W. Embleton	2002
Richard H. Lyon	2003
Chester M. McKinney	2004
Allan D. Pierce	2005
James E. West	2006
Katherine S. Harris	2007
Patricia K. Kuhl	2008
Thomas D. Rossing	2009
Jiri Tichy	2010
Eric E. Ungar	2011
William A. Kuperman	2012
Lawrence A. Crum	2013
Brian C. J. Moore	2014
Gerhard M. Sessler	2015

CITATION FOR WHITLOW W. L. AU

. . . for contributions to understanding underwater biosonar, and for service to the Acoustical Society

SALT LAKE CITY, UTAH • 25 MAY 2016

Whitlow Au received a B.S. degree in Electrical Engineering in 1962 from the University of Hawaii and M.S. and Ph.D. degrees in Electrical Engineering from the University of Washington in 1964 and 1970. His research was specialized on propagation of radio signals in the ionosphere, and in magnetosphere physics. From 1964 to 1968, he worked as a research and development project officer and engineer at the U.S. Air Force Weapons Laboratory in New Mexico, focusing on the passage of radar signals through the plasma sheath that surrounds reentry vehicles. Beginning in 1970, Whit became involved in acoustics as Scientist for the Naval Ocean Systems Center in San Diego, dealing with theoretical aspects of acoustic propagation. He moved to Hawaii in 1971 as Senior Scientist for the Naval Ocean Systems Center, where he did seminal research on dolphin biosonar as well as a wide variety of allied work in underwater sound propagation, acoustic backscatter, digital and analog signal processing, and associated electronics. His 1993 book, *The Sonar of Dolphins*, plus over 50 papers at the time along with numerous meeting abstracts, the majority in the *Journal of the Acoustical Society of America* (JASA), were major products of that period in his professional life. Among these, there is an interesting deviation into Hawaiian katydid sounds in an abstract from 1992 (JASA 92, Pt. 2, 2423) with J. Strazanac. He has published an important study of noise levels in the ocean, *Marine Mammal Populations and Ocean Noise: Determining When Noise Causes Biologically Significant Effects* (The National Academies, 2005). More recently, Whit and Mardi Hastings published their book, *Principles of Marine Bioacoustics*, in 2008.

In 1993, the Navy facility moved to San Diego, and the dolphin work in Hawaii transferred to the Hawaii Institute of Marine Biology, where Whit has served as Senior Scientist since then. During both his San Diego and Hawaii periods, Whit has made a series of major technical and conceptual contributions to Animal Bioacoustics in the study of biosonar in dolphins. In the course of his work, Whit has received the Navy Meritorious Civilian Service Award (3rd highest national award for a Navy civilian employee) for contributions in dolphin bioacoustics in 1986 and Publication Awards from the Naval Ocean Systems Center in 1990, 1991, and 1992. Much of the technical interest in biosonar comes from the underwater sonar concerns of the U.S. Navy, for which Whit has worked on program planning and evaluation for the Office of Naval Research (ONR) and the Ocean Studies Board of the National Research Council for the National Academies.

In 1998, Whit was awarded the Acoustical Society of America Silver Medal in Animal Bioacoustics for his technical contributions to understanding the biosonar of cetaceans. This award specifically recognized his leading role in transforming much of the loosely-structured research about echolocation in cetaceans into a solid, comprehensive body of knowledge that has both scientific and technological importance. His book, *The Sonar of Dolphins* in 1993, and a new historical review in *Acoustics Today*, in the fall 2015 issue, are the best comprehensive and summary entry points for anyone seeking to understand underwater biosonar. In 2007, Whit co-authored a comparative summary of dolphin and bat biosonar for *Physics Today* with James Simmons. During Whit's career, he has published over two hundred scientific papers.

Whit has been an influential leader in the Acoustical Society of America. His service has included being a founder, member (1989-1994), and chair of the Technical Committee on Animal Bioacoustics (1997–2000) and a member of the Executive Council (2001–2004). He was elected Fellow in 1990 and served as Vice President (2006–2007) and President (2009–2010). Since 1998 he has served as an Associate Editor of JASA for Animal Bioacoustics. In his capacity as ASA President, he was also a member of the Council of Scientific Society Presidents. For facilitating scientific exchange, he organized and chaired special sessions in Psychological and Physiological Acoustics or Animal Bioacoustics at ASA meetings in San Diego (1983), Hawaii (1988), Houston (1991), and Austin (1994). He was a member of the organizing committee for the joint Acoustical Society of America/Acoustical Society of Japan (ASJ) meeting held in Hawaii in 1996, ASA Chair of the Joint ASA/ASJ meeting in 2006, and the ASA Chair for Acoustics 2012 Hong Kong.

The intellectual clearinghouse of research on biosonar has been a series of major international conferences held roughly every decade. Whit was a member of the organizing committee for the meeting held on Jersey, Channel Islands (1979), the meeting in Helsingor, Denmark (1986) and the meeting in Kyoto, Japan (2008). The Society has sponsored several large-scale workshops on Acoustic Communication by Animals, including at the University of Maryland and Cornell University. Whit was a co-organizer for the meeting at Oregon State University, Corvallis, in 2008. For marine mammal bioacoustics, he was on the organizing committee and a session chairman for the International Symposium on Sensory Systems and Behavior of Aquatic Mammals, held in Moscow, Russian Federation, in 1991, and on the organizing committee for the 13th Biennial Conference on the Biology of Marine Mammals, held in Wailea, Hawaii, in 1999.

Whit's most lasting act of service to the Society has been to create the Animal Bioacoustics Technical Committee, around which has coalesced much of the Society's current activity in research on the uses of sound by animals. This happened at a critical time in the history of the Society, and it was instrumental in preserving the central role of the Society for representing acoustics in biology. Up until about 1980, the Society was unchallenged as the "go-to" place for research on the sense of hearing and its mechanisms, embodied in the meeting sessions of the Psychological and Physiological Acoustics Technical Committee. Then, a parallel scientific association developed around this area emphasizing the clinical implications of research on hearing, which by 1980 had begun to draw much of this activity away from the Acoustical Society. Whit rescued the central role of the Society when he carved out the Animal Bioacoustics area as a separate entity. This technical committee has continued the large, comparative study of hearing and sound production within the Society. The importance of sound for animals is such that the Society's presence is crucial for its mission representing this area.

In sum, Whit has made outstanding contributions to all of our shared enthusiasms for acoustics at both the technical and the institutional levels, which makes him an excellent recipient of the Society's Gold Medal.

JAMES A. SIMMONS

Session 4aAA

Architectural Acoustics: Emerging Parametric/Generative Design Tools in Architectural Acoustics

Ian B. Hoffman, Chair
Dept. of Architecture, Judson Univ., 1151 N State Street, Elgin, IL 60123

Chair's Introduction—8:30

Invited Papers

8:35

4aAA1. Generative acoustics in architecture. Emily Schilb (Edward Dugger + Assoc., P.A., Stuart, VA), Michael Ermann (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu), David Rife (Dave and Gabe, Blacksburg, VA), and Andrew Hulva (Architecture + Design, Virginia Tech, Blacksburg, VA)

The researchers share three studies in parametric, generative, and iterative architectural acoustics. (1) A real-time parametric model shapes a geometrically complex ceiling plane with grips in order to visualize first-order reflections so that early arriving strong reflections may be directed to the audience and late-arriving echoes may be avoided. (2) A ray-tracing software model pivots door openings so that a coupled volume concert hall can be calibrated to establish the "sweet spot" range of aperture sizes that are more likely to produce a double-sloped sound decay. (3) A custom-designed auralization software simulates sound transmission loss so that designers may hear the (relative) noise isolation of different assemblies.

9:00

4aAA2. Elucidation of acoustical phenomena through the re-tooling of comprehensive acoustical simulations. Arthur W. van der Harten (Acoust. Distinctions, 145 Huguenot St., New York City, NY 10801, Arthur.vanderharten@gmail.com)

In its 8 years available to the general public, the open source acoustical simulation tool Pachyderm Acoustic for Rhinoceros has been used not only for prediction of room quality and noise level, but in a variety of other ways which help to clarify the physical behavior of sound in space. Since a scripting interface for Ironpython was released in 2011, and a grasshopper interface in 2015, Pachyderm has earned a place in the growing field of customizable workflows. Its geometrical and numerical tools have been re-purposed for customized visualizations, real-time analysis, search algorithms, and even sculpture. This talk exhibits some of the most interesting work done by re-tooling Pachyderm Acoustic to-date, and attempts to speculate on how it may be used in the future, as designers with more knowledge of acoustics begin to emerge in the workplace.

9:25

4aAA3. Parametric design applications in architectural acoustics—Generation and optimization of reflective surfaces for specific source/receiver combinations. Marcus R. Mayell (Threshold Acoust. LLC, 141 West Jackson Blvd., Chicago, IL 60604, mmayell@thresholdacoustics.com) and Ian B. Hoffman (Architecture, Judson Univ., Elgin, IL)

This investigation considers the potentials of what are normatively visual parametric design tools, within acoustics design thinking. The typical workflow inherent to most room acoustic software consists of architectural design followed by analysis. In response to the emergence of parametric tools in architectural design, like Rhino and Grasshopper, acoustic designers now have the potential to carry out some analysis through visualization in a more streamlined manner. Furthermore, parametric tools carry the capability to alter the typical workflow from "design followed by analysis" to a more integrated design and analysis workflow, or in some cases even a set criteria followed by a set of design solutions. Through investigation and application of these idea as a part of a recent Master of Architecture thesis project at Judson University, I have been able to develop and test specific parametric definitions which define and visualize reflecting surface potentials based on set source and receiver area criteria.

9:50

4aAA4. Statistical considerations of early-stage design by using computer-based tools in architectural acoustics. Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de) and Shuai Lu (School of Architecture, Tsinghua Univ., Beijing, China)

Room simulation techniques require input data of the room model and the boundary conditions. In an iterative early-stage design process, the general room shape and the boundary conditions correspond to the perceptual parameter space such as reverberation time, strength, clarity, etc. The influence of specific settings in the algorithmic details such as the number of rays, the temporal resolution, the filter bandwidth will be demonstrated by calculating room impulse responses under the condition of variation of such settings. Furthermore, the statistics of the input variables of the general shape result in certain degrees of freedom for the absorption and scattering coefficients. It is discussed how a design space could be defined, which describes the probability for achieving the desired room acoustic performance as a function of the general room shape and boundary conditions.

10:15–10:30 Break

10:30

4aAA5. Lost in translation, easing communication through the use of digital modeling. Shane J. Kanter, Gregory Miller, and Marcus Mayell (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com)

Daily communication between Acoustician, Architect, facility users, building owners, and consultants paves a road riddled with several opportunities for miscommunication. Although many design team members have honed their auditory senses, many of us communicate most effectively through the use of visual graphics and diagrams. Computer aided design, including the use of moving images, has not only become a tool for analysis but also allowed for more effective communication to educate and influence important design decisions. Parametric modeling tools allow consultants to develop analysis and communication tools early on and throughout the design of architectural elements. Two recent applications of the digital modeling process include the new orchestra shell at the Lyric Opera in Chicago and a 1000 Seat Convening Hall. Digital tools used to model and analyze acoustic performance have been imperative while working through the challenges embedded within each project. In both cases, these tools have helped overcome design and communication challenges among the team, leading to a result that meets the unique needs of each space.

10:55

4aAA6. Hope in mirrors: Update on using SketchUp and light rendering to visualize acoustic reflections. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

At a previous ASA meeting, a method was shown that allows architects to visualize acoustics in an intuitive manner using tools already familiar in the design workflow. The method is based upon some of the earliest forms of acoustics modeling, whereby mirror surfaces and light sources were used in scale models to determine the architecture's impact on early reflections (a physical modeling of the image source method, essentially). This talk will demonstrate applications using this revitalized method, as well as expand on the original method using 360-degree immersive imagery, animating discrete reflection orders spatially (as opposed to temporally), and other useful techniques which are afforded by virtual modeling that would never otherwise be permitted by natural physics.

THURSDAY MORNING, 26 MAY 2016 — SALON I, 8:00 A.M. TO 11:00 A.M.

Session 4aAO

Acoustical Oceanography and Animal Bioacoustics: Noise Impacts from the Industrialization of the Outer Continental Shelf and High Seas

Michael A. Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Chair's Introduction—8:00

Invited Papers

8:05

4aAO1. Noise sources from the industrialization of the ocean. Michael Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

Increasingly technology is opening up hostile and challenging marine environments for industrial exploitation. This is occurring in the energy sector with fossil fuel exploration and extraction operations and developing wind and hydrodynamic energy projects. It is also occurring with the deep-water expansion of other extraction industries such as minerals mining and fishing. All of these operations and enterprises are introducing loud and complex noise sources into marine bioacoustic habitats. This presentation will be an overview examination of existing and developing noise sources that are a consequence of the industrialization of the outer continental shelf and high seas.

8:25

4aAO2. Underwater sound radiation from subsea factories. Bas Binnerts (Acoust. and Sonar, TNO, Oude Waalsdorperweg 64, Den Haag 2597AK, Netherlands, bas.binnerts@tno.nl) and Pieter v. Beek (Fluid Dynam., TNO, Delft, Netherlands)

In the Oil & Gas industry, there is a trend toward more subsea activities such as the processing of the Oil and Gas in so-called "subsea factories." In this work, an overview is presented of the various anthropogenic sources contributing to the soundscape during the operational phase of these factories. With the measured sound spectrum (in air) of a complete turbo compressor installation and the known differences between radiation into air and sea water, a stylized, equivalent monopole source level is constructed. This source level is put into perspective by comparing it against a variety of other anthropogenic continuous sources. For a stylized subsea factory, it was judged that the turbo compressor will dominate the generated sound field of subsea factories. Finally, potential risks of the radiated sound are identified and possible sound mitigation solutions are discussed.

8:45

4aAO3. Underwater sound signatures of offshore industrial operations. Christine Erbe, Kim Allen, Alec Duncan, Alexander Gavrilov, Robert McCauley, Iain Parnum, Miles Parsons, and Chandra Salgado-Kent (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

Marine industries, such as offshore petroleum, minerals, fisheries, transportation, tourism, defence, etc., introduce sound underwater, changing marine soundscapes, ranging from shallow coastal to deeper offshore regions. Concern about potential noise impacts on marine fauna has led to numerous underwater recordings and bioacoustic studies. Prior to operations, e.g., as part of permit applications for marine operations, environmental impact assessments are carried out that rely on the modelling and prediction of sound emission, propagation, and impacts. A catalog of sound signatures from activities ranging from exploration and surveying to construction, production, general operation, and decommissioning is necessary for predictive modeling. Underwater sounds recorded from seismic airguns, sub-bottom profilers, echosounders and sonars (sidescan, single-beam, and multi-beam), marine traffic (from small boats to large ships), aerial transportation (helicopters recorded underwater), dredging, pile driving, explosions, drilling, floating petroleum production, storage facilities, etc., are reviewed and their spectral and temporal characteristics, as well as beam patterns are discussed.

9:05

4aAO4. Marine soundscape during a shallow-water seismic survey in the Arctic Ocean. Shane Guan (Office of Protected Resources, NOAA/NMFS, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20910, shane.guan@noaa.gov) and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

For noise generating activity that lasts for an extended period of time, an overall increase in noise levels and change of soundscape within a larger area (over tens of km^2) can be expected. This study analyzed the sound field characteristics during a shallow-water marine seismic survey in the Beaufort Sea of the Arctic Ocean. Three bottom mounted acoustic sensors were deployed in the survey area: two outside the barrier islands in water depths about 12 m, and one inside the barrier islands in water depth of 2.8 m. Averaged 1 min sound pressure levels (SPLs) in broadband, 100–500 Hz, 1–5 kHz, and above 10 kHz bands were computed for periods when airguns were active and inactive. The results showed an 8-dB increase during the period when airguns were active in the 100–500 Hz band for two locations outside the barrier islands. However, there was no noticeable difference in SPLs during periods airguns were active and inactive inside the barrier islands. This is probably due to higher natural ambient noise and low-frequency cut-off of airgun pulses in this extreme shallow location.

9:25

4aAO5. Underwater sound directionality of commercial ships. Martin Gassmann, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0205, mgassmann@ucsd.edu)

The underwater sound radiated by commercial ships is an unintended by-product of their operation and one of the most significant contributors to man-made noise at low frequencies in the ocean. To estimate the directionality of underwater sound radiated by current commercial ships, a seafloor array of five high-frequency acoustic recording packages (HARPs) deployed to 1 km depth with a maximum horizontal aperture of 1 km was used. As a ship of opportunity passed over the HARP array, the directions from the ship to each HARP along with the corresponding source levels were estimated for each ship location. Ships were tracked via satellites (Automatic Identification System—AIS) and acoustically by a frequency domain beamformer that was implemented for one of the HARPs configured with a volumetric hydrophone array (2 m maximum aperture). The directionality estimates of contemporary commercial ships exhibit significant stern-bow asymmetries among other quantitative characteristics that will be discussed.

9:45–10:00 Break

10:00

4aAO6. Comparing methods for estimating the injury and behavioral disturbance radii from sound source characterization measurements. Bruce Martin, Jeff MacDonnell (JASCO Appl. Sci., 32 Troop Ave., Ste. 202‘, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), Alexander O. MacGillivray, and David Hannay (JASCO Appl. Sci., Victoria, Br. Columbia, Canada)

It is a common practice for regulators to require project proponents to estimate the radius around a sound source where marine life could be injured or disturbed. Acoustic propagation modeling is normally required so that realistic radii are obtained that take into account the bathymetry, bottom properties, water column sound velocity profile, as well as source spectrum and directivity. Increasingly proponents are also asked to perform *in-situ* sound source characterization (SSC) measurements to verify the modeling predictions, including the spectrum and directivity of the sound source. During an SSC, sound levels are measured at increasing range from the source. In most cases, it is not possible to measure the sound levels at all ranges of interest, and therefore, the data must be interpolated or extrapolated to estimate the radii of the regulatory sound isopleths. This talk uses real-world data to identify the strengths and weaknesses of four methods of estimating the radii: (1) the practical spreading model; (2) linear interpolation; (3) linear regression with absorption; and (4) model-measured fits.

10:20

4aAO7. The effect of operational measures on shipping sound in the North Sea. Bas Binnerts (Acoust. and Sonar, TNO, Oude Waalsdorperweg 64, Den Haag 2597AK, Netherlands, bas.binnerts@tno.nl) and Christ de Jong (Acoust. and Sonar, TNO, The Hague, Netherlands)

From studies into marine biology, it is known that the behavior of marine mammals and fish can be influenced by the ambient underwater sound level. In busy seas like the North Sea in Europe, sound from shipping traffic is largely responsible for the low frequency part of the ambient sound. In this work, possibilities to regulate the underwater radiated sound of ship traffic are investigated. This involves measures that look at traffic flows and operational use of the vessels, to determine what can be done to reduce the shipping sound levels in certain marine areas. Three types of operational measures are considered: (1) the effect of spatial planning (for example, changing the location of shipping routes), (2) the introduction of a radiated sound limit for vessels in a certain area, and (3) the introduction of a speed limit for vessels in a certain area. The proposed measures have been analyzed on their effect and effectiveness, by means of numerical analysis. Calculations have been performed for a generic ship traffic flow, based on actual recorded AIS data from a North Sea shipping lane, using a speed depended model for the source level of different ship types and a sound propagation model.

10:40–11:00 Panel Discussion

THURSDAY MORNING, 26 MAY 2016 SALON B/C, 8:00 A.M. TO 11:45 A.M.

Session 4aNS

Noise, ASA Committee on Standards, Animal Bioacoustics, Engineering Acoustics, and Physical Acoustics: Wind Turbine Noise I

Nancy S. Timmerman, Cochair
P.E., 25 Upton Street, Boston, MA 02118

Kenneth Kaliski, Cochair
RSG Inc., 55 Railroad Row, White River Junction, VT 05001

Robert D. Hellweg, Cochair
Hellweg Acoustics, Wellesley, MA 02482

Paul D. Schomer, Cochair
Schomer and Associates, Inc., 2117 Robert Drive, Champaign, IL 61821

Chair's Introduction—8:00

Invited Papers

8:05

4aNS1. A case study in Canada: Exploring research challenges of industrial wind turbines and health. Carmen M. Krogh (Killaloe, Killaloe, ON K0J 2A0, Canada, carmen.krogh@gmail.com) and Jeffery Aramini (Fergus, ON, Canada)

The topic of adverse health effects associated with industrial wind turbines (IWT) is controversial and debated worldwide. Some residents living in proximity to wind energy facilities report symptoms of sleep disturbance, annoyance, headaches, ear pain/discomfort, mood disorders, stress, cardiac and blood pressure effects, reduced quality of life, and other adverse effects. In some cases, research initiatives have been the result of individuals' complaints. The research is challenged in part by the complexities and numerous variables associated with this subject. A range of IWT research approaches, sometimes in combination with each other, has been used including self-reporting surveys, investigations and acoustical measurements. On July 12, 2012, Health Canada announced its large scale cross-

sectional randomized epidemiological wind turbine noise and health study. The research was conducted by Health Canada in two Canadian provinces. Health Canada indicated its study had limitations, would not be definitive, and would not permit any conclusions to be made with respect to causality. This presentation will explore some of the inherent challenges of studying health effects associated with wind energy facilities and will consider the role of those individuals reporting adverse health effects.

8:25

4aNS2. Industrial wind turbines and adverse health effects: Where we are, where we need to go, and the need for regulations and predictive models to recognize human physiology. Michael A. Nissenbaum (Radiology, McGill Univ., 1 Westmount Square, Ste. C210, Westmount, QC H3Z 2P9, Canada, mnissenbaum@att.net)

Since our initial study was published (Nissenbaum *et al*., 2012), work in several areas of human physiology has begun to elucidate the precise mechanisms by which sleep disturbances result in adverse health effects, over both short and longer durations. These include impaired neuronal connections in the learning brain, altered genetic expressions impacting the immune system, and correlations between poor quality sleep and MRI-measured atrophy of the brain over a mean period of 3.5 years. Additionally, fMRI has demonstrated brain responses to sounds with frequencies as low as 8 Hz. At lower frequencies, somatosensory mechanisms are now thought to play a role, in addition to auditory. Local regulations regarding noise (Soundscape) limits and methods of measurement were designed prior to current understandings of human sensory and reactive physiology. Instrumentation and modeling geared towards satisfying those regulations are by implication lacking because they do not capture or predict physiological responses to IWT noise. According to the principles of soundscape, and given the subtleties of human physiology, humans remain the best instruments available for detecting objectionable noise and identifying adverse health effects. Regulations, measurement methods, and predictive models must adapt to current understandings of human physiology to best protect human populations.

8:45

4aNS3. Measuring perceptual effects of infrasound. Peggy B. Nelson, Michael Sullivan, Meredith Adams, Matthew Lueker, Jeffrey Marr (Univ. of Minnesota, 164 Pillsbury Dr. Se, Minneapolis, MN 55455, peggynelson@umn.edu), and Bruce Thigpen (Eminent Technologies, Tallahassee, FL)

A multi-disciplinary group of researchers at the University of Minnesota Center for Applied and Translational Sensory Science (cats-s.umn.edu) are designing and pilot testing the perceptual effects of infrasound, in collaboration with Eminent Technologies (rotarywoofer.com). An infrasound generator simulates the acoustic signature of audible sound and infrasound generated by wind turbines in the field. With the support of Xcel Energy, the team of engineers, otologists, hearing scientists, and balance experts are evaluating the effects of infrasound only, acoustic turbine sound, and combined infrasound and acoustic sound. We are testing listeners' quiet detection, masked detection, discrimination, and rating of signals. Pilot results will be described. Together we hope to test the range of perceptual responses to turbine-generated infrasound and audible sound. [Work supported by Xcel Energy RD4-12 to Jeffrey Marr, St. Anthony Falls Laboratory.]

9:05

4aNS4. Effects caused by sounds at low frequencies. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de), Brigitte Schulte-Fortkamp (TU Berlin, Inst. of Fluid Mech. and Eng. Acoust., Berlin, Germany), Wade Bray (Acoust., Inc., Brighton, MI), and Klaus Genuit (Acoust. GmbH, Herzogenrath, Germany)

A central tenet of the Soundscape concept is that humans immersed in sonic environments are as "new experts" reliable communicators, whose reports and descriptions of sound and its effects must be taken seriously. As a first step, different experiments focusing on wind turbines coupled with detailed questionnaires were carried out to study the link between specific characteristics of wind turbine noises and annoyance reactions. It seems as if this will enable to determine the benefit of the Soundscape approach for investigated low frequency noise phenomena. The challenge of validating and resolving widespread and growing reports of health, physiological and behavioral effects apparently from new environmental infrasound can be met through coupling soundscape with parallel scientific techniques not limited to acoustic measurements. The soundscape concept has an incipient role to connect and approach the resolution of this complex issue. Such an augmented soundscape approach centering on the sensitivity of human beings is as important and applicable to responses to effects from sound as it is to responses to directly audible sound. This is a new kind of merged sound quality and public health issue combining acoustic, psychoacoustic, and medical aspects. Its resolution depends on a combined soundscape-mediated approach.

9:25

4aNS5. Patient Centered Medicine and Soundscape—A bridge between clinicians and acousticians. Robert Y. McMurtry (Surgery, Western Univ., 403 Main St., Picton, ON K0K2T0, Canada, rymcmurtry1@gmail.com)

The Patient Centred Method (PCM) and Soundscape have much in common including their emergence about 60 years ago based on the work of Balint and Kryter, respectively. Both place the patient or person at the center of management of clinical illness or noise annoyance. PCM requires that the patient perceive that they have experienced meaningful care, communication, and common ground in clinical encounters. The evaluation focuses on the patient's life context and their perception of disease or the "illness experience." When PCM is accomplished the result is higher satisfaction, better outcomes of chronic diseases, fewer tests and referrals, and attendant lower costs (Stewart *et al*. 2000). Soundscape, a term coined by Shafer in 1977 also places the person in center, in the context of their sonic environment, emphasizes their perception of noise as the "New Experts" (Bray 2012). According to Bray, exposed people are "objective measuring instruments whose reports and experiences must be taken seriously and quantified by technical measurements." This paper will explore the congruence of PCM and Soundscape and the applicability to environmental noise assessments. The necessity of this approach in evaluating the impact (e.g., PTSD) on those exposed to wind turbine acoustical energy will be explored.

9:45

4aNS6. Threshold of hearing versus threshold of sensation for low frequency and infrasound. Steven E. Cooper (The Acoust. Group, 22 Fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au)

Residents impacted by wind farms identify the perception of a sensation not necessarily noise. By use of a pressure sound field in a damped reverberation chamber the Threshold of Hearing for Infrasound and Low Frequency (refer Wattanbe and Moller) has been explored together with the Threshold of Sensation for sine waves. Those results are then compared using band limited pulsed signals from a Wind Farm. The differences and what they mean for Wind Turbine investigations are discussed.

10:05–10:25 Break

10:25

4aNS7. Wind farm infrasound—Are we measuring what is actually there or something else? (Part 2). Steven E. Cooper (The Acoust. Group, 22 fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au)

In predigital acoustics, low frequency analysis used analog narrow band filters and cathode ray oscilloscopes for special problems leading to the general use of peak values. Analog filters have time constants that can affect the derived rms values requiring caution where high crest factors are involved. Modern narrowband digital analysis based on a FFT of the time signal extracts the periodic function that occurs in the time domain that are then displayed as discrete peaks in the frequency domain. FFT analysis of turbines show discrete infrasound peaks at multiples of the blade pass frequency in addition to sidebands in the low frequency range spaced at multiples of the blade pass frequency. Are these signals actually there or are they a product of modern day analysis. Is the infrasound signature a clue to a different area of investigation? In Jacksonville, Part 1 presented the complexity of the investigations and showed how the raw Pascal data are lost when converting to SPL and then A-weighting. Part 2 presents the results of different filtering techniques for different wind farms.

10:45

4aNS8. An evaluation of how nightly variations in wind turbine noise levels influence wrist actigraphy measured sleep patterns. David S. Michaud (Health Canada, Canadian Federal Government, 775 Brookfield Rd., Ottawa, ON K1A 1C1, Canada, david.michaud@canada.ca)

Health Canada's Wind Turbine Noise and Health Study assessed self-reported and objective measures of sleep on a sub-sample of the study's 1238 participants. The data analysis indicated that calculated long term outdoor wind turbine noise (WTN) levels up to 46 dBA did not have a significant influence on the evaluated measures of sleep (Michaud *et al*., 2016, Sleep **39**, 97–109). A more refined analysis is being conducted to assess wrist actigraphy measured sleep patterns in relation to nightly variations in wind turbine operations. Variations in turbine operations (i.e., RPM and electrical power) were used to calculate WTN levels in 10-min intervals and time-synchronised with sleep watch data collected in 1-min epochs. The 10-min sound levels and daily sleep diaries (relied upon to adjust for closed or open windows) were used to estimate indoor A-weighted WTN levels. The correction factor used to obtain indoor sound levels was derived from a series of field measurements designed to investigate the outdoor to indoor sound pressure level difference on a representative sample of dwellings. The analysis is restricted to participants living between 0.25 and 1 km from wind turbines (116 males and 159 females). At these distances, WTN levels are the highest making it more likely to detect potential WTN impacts on sleep. Results will be presented for multiple measures of sleep in 10-min intervals and nightly averages for up to seven consecutive sleep nights.

11:05

4aNS9. Wind turbine acoustic analysis should be performed to minimize the potential for adverse impacts on life-quality for neighbors. Robert W. Rand (Rand Acoust., 1085 Tantra Park Circle, Boulder, CO 80305, rrand@randacoustics.com) and Stephen E. Ambrose (S.E. Ambrose & Assoc., Windham, ME)

Wind turbine environmental assessments should be expanded to include evaluating infrasonic, low-frequency, and A-weighted noise levels. Modern instruments and microphones can measure three simultaneous, band-pass measurements: dBL, dBC, and dBA. This paper will evaluate these band-pass measurements and discuss potentials for predicting potential adverse public responses.

11:25

4aNS10. Reproducing wind farm infrasound for subjective testingJust how accurate is the reproduced signal? Steven E. Cooper (The Acoust. Group, 22 Fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au)

Apparently on the basis of room modes being excited in residential dwellings, one concept has been to ignore high quality field measurement recordings (wave files) and use the narrow band (FFT) Leq results of a 10 min sample to create, by superimposing a number of sine waves, a trapezoidal time signal as the source for subjective testing and restricting the bandwidth to only infrasound. Other testing utilizes full spectrum signals but has limitations on having an accurate signal due to the limitation of the speakers. An examination of both methods and the limitations of the results have been examined and will be discussed.

Session 4aPA

Physical Acoustics: Multiple Scattering I

Valerie Pinfield, Cochair
Chemical Engineering Department, Loughborough University, Loughborough LE11 3TU, United Kingdom

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

Chair's Introduction—8:00

Invited Papers

8:05

4aPA1. Integrated extinction, or total scattered energy, as a linear sum of individual contributions in acoustic multiple scattering. Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Multiple scattering (MS) involves interaction between scatterers with the result that the scattered field is not simply the sum of the separate responses. In fact, it is generally a very nonlinear process, particularly at high frequencies. It is therefore surprising that there are circumstances for which the total energy of the scattered acoustic wave is simply the sum of the individual contributions. Total scattered energy, also known as the integrated extinction, is defined as the integral of the scattering cross-section with respect to wavelength for a given direction of incidence, and is therefore a measure of the scattering over all frequencies. Two conditions are necessary, one at zero frequency and the other at infinite frequency. They are, respectively, (i) that all scatterers have effective static compressibility and density equal to the background properties, and (ii) that the wavefront in the forward direction associated with MS does not precede the incident wavefront. Examples will be given showing how both conditions can be achieved with regular and irregular arrays of elastic shells. This linear MS result is particular to acoustics, as compared with electromagnetic waves *in vacuo*. [Work supported by ONR.]

8:25

4aPA2. The effective material properties of a composite elastic half-space. Ian D. Abrahams and William J. Parnell (School of Mathematics, Univ. of Manchester, Manchester M13 9PL, United Kingdom, i.d.abrahams@manchester.ac.uk)

A classical problem in applied mathematics is the determination of the effective properties of a composite material by looking at its reflection and transmission properties. This model discussed here is an elastic half-space containing randomly distributed voids—to obtain its "average" material properties (i.e., the effective density and elastic moduli) we consider elastic waves incident from a homogeneous half-space onto the inhomogeneous material. We restrict attention to dilute dispersions of inclusions, and therefore, results are obtained under the assumption of small volume fraction. We look at several aspects of this problem, as times allows. First, we discuss how predictions derived from the non-isotropic Foldy or the Waterman-Truell multiple-scattering theories (MSTs) in the low-frequency limit are equivalent to results found by an asymptotic integral equation (homogenization) method developed by the authors [1]. Second, the effect of nonlinear pre-stress on each void, and hence on the averaged material properties, is considered [2]. And third, some comments are offered regarding higher order effects and the closure assumption (e.g., the quasi-crystalline approximation). [1] W. J. Parnell, I. D. Abrahams, and P. R. Brazier-Smith, Quart. J. Mech. Appl. Math. **63**, 145–175, 2010. [2] T. Shearer, W. J. Parnell, and I. D. Abrahams, **471**, 20150450, 2015.

8:45

4aPA3. Full transmission and reflection of waves propagating through a maze of disorder. Benoît Gérardin, Jérôme Laurent, Arnaud Derode, Claire Prada, and Alexandre Aubry (ESPCI ParisTech, PSL Res. Univ., CNRS, Université Paris Diderot, Sorbonne Paris Cité, Institut Langevin, 1 rue Jussieu, Paris 75005, France, benoit.gerardin@espci.fr)

Multiple scattering of waves in disordered media is often seen as a nightmare whether it be for communication, imaging, or focusing purposes. The ability to control wave propagation through scattering media is thus of fundamental interest in many domains of wave physics, ranging from optics or acoustics to medical imaging or telecommunications. Thirty years ago, it was shown theoretically that a properly designed combination of incident waves could be fully transmitted through (or reflected by) a disordered medium. Although this remarkable prediction has attracted a great deal of attention, open and closed channels have never been accessed experimentally. Here, we study the propagation of elastic waves through a disordered elastic waveguide. Thereby, we present experimental measurements of the full S-matrix across a disordered elastic wave guide. To that aim, laser-ultrasonic techniques have been used in order to obtain a satisfying spatial sampling of the field at the input and output of the scattering medium. The unitarity of the S-matrix is investigated and the eigenvalues of the transmission matrix are shown to follow the expected bimodal distribution. Full transmission and reflection of waves propagating through disorder are obtained for the first time experimentally. The wave-fields associated to these open and closed channels are imaged within the scattering medium to highlight the interference effects operating in each case.

9:05

4aPA4. Stochastic theory for multiple scattering of ultrasound in particulate media. Felix Alba (Felix ALBA Consultants, Inc., 1159 Sunset Dunes Way, Draper, UT 84020, felixalba@q.com)

Even though there have been published/used several approaches to modeling the complex phenomenon of *multiple scattering*, they are either of a *semi-empirical* nature with severe limitations in size/wavelength range and performance or, albeit having a *fundamental flavor*, have proven not accurate enough to be employed in broad range/application scientific instruments. Fundamental versus phenomenological approaches are discussed. The METAMODEL^{TM} fundamental multiple-scattering theory is as fundamental as Mie (optics) and ECAH (acoustics) single-scattering theories. Besides its *fundamental* character, its *generic* validity resides in having described and calculated the detailed interaction between the scattered fields produced by every particle, treating such an interaction in a statistical sense and leading to *generic stochastic field equations* where the overlapping of all scattered fields is equally contemplated regardless of their physical nature (viscous-inertial, thermal diffusion, elastic, electromagnetic, etc.). The fundamentals of this theory are described in some mathematical detail. Using the Ultra-SCATTERER^{TM} R & D software tool, predicted as well as experimental data in concentrated suspensions, emulsions, and aerosols (collected with commercially available spectrometers) are presented.

9:25

4aPA5. Acoustic propagation and multiple scattering in powders. Malcolm J. Povey, Mel Holmes, and Raied Al-Lashi (School of Food Sci. and Nutrition, Univ. of Leeds, Leeds, West Yorkshire ls2 9jt, United Kingdom, m.j.w.povey@leeds.ac.uk)

Determination of the material properties of powders is critical in very many industries yet crucial transformations such as the transition from a free-flowing powder to a caked powder defy quantitative measurement. In particular, distinguishing between the onset of caking (so that it may be prevented, for example) and consolidation or segregation of powders which have not caked is very difficult. We present here calculations of the acoustic properties of free flowing powders based on scattering theory (Alba, 2004), together with model calculations for caked and consolidated powders within which an effective stiffness has developed (Coghill 2011, Makse 2004). These calculated values are compared with the measured acoustic properties of silica powders demonstrating that acoustic measures such as the velocity of sound and the attenuation of sound (insertion loss) are highly effective in distinguishing between caking and consolidation. Alba. 2004, Chapter 10 of "Concentrated dispersions, theory, experiments, and applications," edited by P. Somasundaran and B. Markovic, American Chemical Society (ACS) Symposium Series 878. Coghill and Giang, Powder Technol. **208**, 694–701, 2011. doi:10.1016/j.powtec.2010.11.040. http://dx.doi.org/10.1016/j.powtec.2010.11.040. Makse, Gland, Johnson, and Schwartz, Phys. Rev. E**70**, 1–19, 2004. doi:10.1103/PhysRevE.70.061302.

9:45–10:05 Break

Contributed Papers

10:05

4aPA6. Nonlinear acoustic forces acting on inhomogeneous fluids at slow time-scales. Jonas T. Karlsen (Dept. of Phys., Tech. Univ. of Denmark, DTU Phys., Bldg. 309, Kongens Lyngby 2800, Denmark, jonkar@fysik.dtu.dk), Per Augustsson (Dept. of Biomedical Eng., Lund Univ., Lund, Sweden), and Henrik Bruus (Dept. of Phys., Tech. Univ. of Denmark, Kongens Lyngby, Denmark)

We present a novel theory describing the nonlinear acoustic force density acting on a fluid of inhomogeneous density and compressibility, for example, due to an added salt concentration. We derive an expression for the time-averaged acoustic force density acting on an inhomogeneous fluid, which depends on the gradients of the fluid density and compressibility. This smeared-out force density can be interpreted as a generalization of the well-known acoustic forces acting on a particle or an immiscible-fluid interface. The special case where the speed of sound in the solution is independent of the salt concentration, which is a good approximation for many actual salt solutions, leads to a particularly simple theoretical description. The theory predicts that in microfluidic channels, the nonlinear acoustic forces act to relocate density distributions into field-dependent configurations, which are stabilized against gravitational collapse driven by hydrostatic pressure gradients. We show the first experimental confirmations of these predictions obtained by confocal imaging in glass-silicon microchips.

10:20

4aPA7. Multiple scattering in bubbly media: Highlighting the role played by interactions between neighboring bubbles. Maxime Lanoy (Institut Langevin, 1, rue jussieu, Paris 75005, France, maxime.lanoy@espci.fr), Valentin Leroy (Matière et Systèmes Complexes, Paris, France), and Arnaud Tourin (Institut Langevin, Paris, France)

A bubble in water exhibits a low-frequency monopolar acoustic resonance, the so-called Minnaert resonance, which makes an assembly of such bubbles an ideal system to study multiple scattering of ultrasound. Following Foldy's seminal work, various approaches, such as the ones by Keller or Waterman and Truell, have been developed to infer the effective acoustic properties of a bubble cloud. Here, we confront the predictions of these different approaches with numerical results obtained with a multiple scattering theory (MST) code that fully incorporates the multiple scattering effects. Based on this study, we show the importance to take into account the interactions between neighboring bubbles to predict the behavior of a bubble sample. Finally, we demonstrate how the introduction of a local order can affect the effective parameters and allows interesting transmission properties.

10:35

4aPA8. Acoustic harmonic generation and phase conjugation with a single layer of bubbles. Olivier Lombard (Univ. Paris Diderot, 10 rue Alice Domon et Leonie Duquet, Paris 75205 Cedex 13, France, olivier.lombard@univ-paris-diderot.fr), Christophe Barriere (Institut Langevin, Paris, France), and Valentin Leroy (Matière et Systèmes Complexes, Paris, France)

Bubbles are well known for being strong nonlinear scatterers in acoustics. This feature is actually used in medical ultrasound, and nonlinear phenomena, such as phase conjugation [1] or second-harmonic generation [2], had been reported in bubbly liquids. We carried out experiments and calculations on a particularly simple bubbly medium: a single layer of bubbles trapped in a yield-stress fluid. We show the existence of an ideal bubble concentration that maximizes the nonlinear second harmonic generation by the layer. It results from the interplay between the nonlinear response and the strong multiple scattering in the system [3]. The same nonlinear mechanism can be used to obtain phase conjugation, with a probe wave at frequency f, and pump wave at 2f. The single layer of bubbles is then a subwavelength phase conjugation mirror. We studied its efficiency in terms of direction and magnitude of the reflected wave. [1] D. V. Vlasov *et al*., Sov. Phys. Acoust. **29** (1983). [2] J. Wu, Z. Zhu, J. Acoust. Soc. Am **89** (6) (1991) [3] O. Lombard *et al*., EPL, **112** (2015).

10:50

4aPA9. Manipulating air bubbles with secondary Bjerknes forces. Maxime Lanoy (Institut Langevin, 1, rue jussieu, Paris 75005, France, maxime.lanoy@espci.fr), Caroline Derec (Matière et Systèmes Complexes, Paris, France), Arnaud Tourin (Institut Langevin, Paris, France), and Valentin Leroy (Matière et Systèmes Complexes, Paris, France)

Gas bubbles in a sound field are submitted to a radiative force, known as the secondary Bjerknes force. In this presentation, we propose an original experimental setup that allowed us to investigate in details this force between two neighboring bubbles. The dependance with the sonication frequency, as well as the bubbles radii and distance, was examined. We report the observation of both attractive and, more interestingly, repulsive Bjerknes force, when the two bubbles have different radii and can thus be driven in antiphase. On the contrary, our results also show the importance of taking multiple scattering into account when the bubbles radii become similar. The setup demonstrates the accuracy of secondary Bjerknes forces for attracting or repealing a bubble, and could lead to new acoustic tools for non-contact manipulation in microfluidic devices.

11:05

4aPA10. Nonstructural acousto-injection luminescence in metalized lithium niobate. Igor Ostrovskii (Phys. and Astronomy, Univ. of MS, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu), Oleg Korotchenkov, Nikolaj Borovoy, Andriy Nadtochiy, Roman Chupryna (Taras Shevchenko Univ., Kyiv, Ukraine), and Chandrima Chatterjee (Phys. and Astronomy, Univ. of MS, Oxford, MS)

The observation of a nonstructural acousto-injection luminescence (NAIL) from metallized $LiNbO_3$ wafers is reported. The X- and Y-cut plates with linear dimensions of a few mm and silver paste electrodes on opposite surfaces are investigated. The experiments are done at room temperature. The fundamental shear modes are excited at MHz-frequencies. We measure the spectra of NAIL, acousto-electric resonance/antiresonance properties, X-ray diffraction rocking curves, acoustic emission accompanying NAIL, and photoluminescence. The NAIL and associated effects appear above a certain threshold acoustical strain of $e = 10^{-5}$. The results are explained in the terms of considerable piezoelectric fields, yielding the charge injection from the metal contacts into crystal along with the strong mechanical stresses leading to dislocations motion. The acoustic emission and X-Ray rocking curves disclose the dislocation motion under $e > 10^{-5}$. The involvement of the microstructural non-uniformities in the effects observed is experimentally identified by the X-ray rocking curves taken at different ultrasound amplitudes and photo-luminescence spectra taken from the different micro-regions of samples. Photoluminescence reveals the charged point defects that may promote an electrical conduction. The distribution of crystal defects along wafers is not uniform, and has a quasi-periodical component with tens to hundreds of microns spacing between their extremal locations.

THURSDAY MORNING, 26 MAY 2016 — SALON F, 8:00 A.M. TO 11:30 A.M.

Session 4aPP

Psychological and Physiological Acoustics: Temporal Aspects of Auditory Processing (Poster Session)

Magdalena Wojtczak, Chair
Psychology, University of Minnesota, 1237 Imperial Ln., New Brighton, MN 55112

All posters will be on display from 8:00 a.m. to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:45 a.m., and authors of even-numbered papers will be at their posters from 9:45 a.m. to 11:30 a.m.

Contributed Papers

4aPP1. Effects of age and hearing loss on coding frequency and amplitude modulation. Kelly L. Whiteford (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu), Heather A. Kreft (Otolaryngol., Univ. of Minnesota, Minneapolis, MN), and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Is phase-locking to temporal fine structure (TFS) selectively affected by hearing loss or ageing? This question has clinical relevance because audiometric measures may be insensitive to TFS deficits linked to auditory neuropathy/dyssynchrony and synaptopathy (“hidden hearing loss”). Previous work has suggested that slow-rate FM is coded via phase-locking to TFS, whereas fast-rate FM is converted to amplitude modulation (AM) via cochlear filtering. This hypothesis was tested by correlating performance in slow- and fast-rate FM with performance in tasks known to reflect TFS coding (interaural-time-difference detection, ITD) and cochlear filtering (forward masking patterns). Subjects with clinically normal hearing at low frequencies between the ages of 20 and 80 years were tested, with approximately 10 subjects per decade. Effects of low- and high-frequency hearing loss were also examined, while controlling for age. Although correlations were found between all measures of FM and AM, no clear specific effect of age was found on TFS coding (independent of AM coding). Preliminary results with hearing-impaired listeners suggest an effect of cochlear filtering that may not be specific to the fast-rate FM detection. Overall, the results do not provide strong evidence for TFS-specific deficits with either age or hearing loss. [Work supported by NIH grant R01DC005216.]

4aPP2. Spike-timing and mean-rate coding of the temporal fine structure and envelope cues in real words. Ian C. Bruce and Michael R. Wirtzfeld (Elec. and Comput. Eng., McMaster Univ., Rm. ITB-A213, 1280 Main St. W, Hamilton, ON L8S 4K1, Canada, ibruce@ieee.org)

A number of studies over the past decade have argued for the importance of temporal fine structure (TFS) cues for the perception of consonants. However, recent investigations indicate that TFS cues from consonants may largely be converted into envelope (ENV) cues by narrowband cochlear filtering, such that these cues are conveyed by the mean-rate response of auditory nerve fibers rather than spike-timing cues. However, these studies used nonsense VCV syllables, and this result may not generalize to real words in which the patterns of ENV and TFS cues may be substantially different, and in which lexical context may play a role. In this study, we used a computational model of the auditory periphery and neural-based speech intelligibility predictors to investigate the TFS and ENV representation of real words from the NU-6 database. Spike-timing and mean-rate cues were evaluated for "auditory chimaeras" created from this database, in which the TFS of one signal is mixed with the ENV of another. The results indicate that the chimaera processing has a bigger impact in general on the mean-rate representation of phonemes than on the spike-timing representation, and inclusion of the spike-timing cues gives better predictions of phoneme perception. [Funded by NSERC of Canada.]

4aPP3. Cues in detection of rhythmic irregularities. Magdalena Wojtczak (Psych., Univ. of Minnesota, 1237 Imperial Ln., New Brighton, MN 55112, wojtc001@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

A recent study [Hove *et al.* (2014). *Proc. Natl. Acad. Sci.* **111**, 10383–10388] showed that the perception of rhythm carried by simultaneously presented low-pitch and high-pitch complex tones is more easily disrupted by timing irregularities in the low tone than in the high tone. This result led the authors to conclude that low-pitch tones are used for laying down the rhythm in music because they are more precise rhythmic time markers than high-pitch tones. In this study, no difference was found in the temporal acuity for low and high tones. Instead, asynchrony detection was found to be better for low-leading than for high-leading tone pairs, regardless of which tone was rhythmically irregular in a sequence. The results are consistent with an asymmetry in the perception of asynchrony between a low and high tone pair without the rhythmic context [Wojtczak *et al.* (2012). J. Acoust. Soc. Am. **131**, 363–377]. The outcome leads to a reinterpretation of the results of Hove *et al.*, based on asynchrony, rather than rhythm, perception. Possible ecological reasons for the asymmetry in asynchrony perception include the natural properties of both sound sources and the peripheral auditory system. [Work supported by NIH grant R01 DC005216.]

4aPP4. Psychophysiological responses to listening to speech in intermittent noise. Alexander L. Francis (Purdue Univ., SLHS, Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu)

Listening to speech in noise can be effortful, and the presence of background noise may in itself provoke physiologically measurable stress. Typical laboratory tests of speech perception in noise often present masked stimuli in trials separated by silence, in effect starting and stopping an otherwise constant background noise multiple times over the course of an experiment. While these interruptions might reduce listener stress by providing momentary respite from an aversive stimulus, intermittent noise might also increase stress by repeatedly inducing automatic physiological responses to the noise onsets. Here, I present the results of an experiment contrasting these possibilities. Younger (age 18–36) and older (age 60+) listeners heard sentences presented in speech-shaped noise at 0 dB SNR while physiological responses linked to stress and arousal (skin conductance, heart rate, fingertip pulse amplitude, and facial electromyography) were recorded. Two roughly 15-min blocks of noise, each containing 36 unique sentences, were presented. In the interrupted noise condition the noise was silenced for 5 s shortly after each sentence while in the uninterrupted condition the noise continued unabated. Behavioral measures of listening task performance and physiological measures collected during listening and speaking will be presented, and implications for future research will be discussed.

4aPP5. The role of a temporal mechanism in the perception of speech-like logarithmic frequency sweeps. Carolyn M. McClaskey (Cognit. Sci., Univ. of California, Irvine, 4308 Palo Verde Rd., Irvine, CA 92617-4321, carolyn.mcclaskey@gmail.com), Daniel Cramer (Oberlin College, Oberlin, OH), and Kourosh Saberi (Cognit. Sci., Univ. of California, Irvine, Irvine, CA)

The auditory system is thought to process dynamic changes in frequency via two complementary mechanisms: a phase-locking-based mechanism that is limited to slowly changing stimuli at low (<4–5 kHz) frequencies, and an energy-based mechanism that functions across all frequency regions and all rates of change. The current study investigates the relative contribution of these two mechanisms—and the role of phase-locking cues in particular—in the perception of low, slow frequency sweeps that closely parallel those found in speech prosody. Listeners identified the direction of unidirectional logarithmic frequency sweeps in a single-interval identification task. Both the rate and extent of frequency change (i.e. transition span) were uniformly varied over a range of 0.0147–0.1667 octaves/s and 0.1–0.5 semitones, respectively. Stimuli were roved around a center frequency of 500 Hz, and all were between 50 and 1000 ms in length. Results show that sensitivity (d') significantly increases with increasing transition span but significantly decreases with increasing rate of frequency change, suggesting that phase-locking cues may contribute to the perception of sweeps with very slow rates of change. Data were well-predicted by a multivariate linear model as a joint function of sweep rate and transition span.

4aPP6. Limitations on temporal processing by cochlear implant users. Robert P. Carlyon, Stefano Cosentino, John M. Deeks (Medical Res. Council, Cognition & Brain Sci. Unit, MRC CBU, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, bob.carlyon@mrc-cbu.cam.ac.uk), Wendy Parkinson (Speech & Hearing Sci., Univ. of Washington, Seattle, WA), and Julie A. Bierer (Speech & Hearing Sci., Univ. of Washington, Washington, WA)

Two experiments studied the deterioration in rate discrimination and pitch perception for pulse trains presented to single electrodes at high pulse rates. The first measured rate discrimination DLs ("RDLs") for 100-pps and 400-pps standard rates, for each of 4–5 electrodes and for 10 Advanced Bionics cochlear implant users. Thresholds were measured using two interleaved adaptive tracks, corresponding to the 100- and 400-pps standard rates. Gap detection thresholds ("GDTs") for a 1031-pps pulse train were also measured. There was a highly significant across-subject correlation between GDT and the 400-pps but not the 100-pps RDL, and these two correlations differed significantly from each other. Similarly, the across-electrode correlation between GDT and the 400-pps RDL was marginally significant, whereas there was no correlation between GDT and the 100-pps RDL. These findings are consistent with the deterioration in high-rate temporal processing sharing a common basis with the mechanisms involved in gap detection, but not with the limitations in low-rate temporal processing. We will also report the results of a second experiment that measured rate discrimination and pitch ranking at low and high rates, both on the same day that patients' implants were activated and after two months of listening experience.

4aPP7. Deficits in the sensitivity to pitch sweeps by school-aged children wearing cochlear implants. Mickael L. Deroche (Ctr. for Res. on Brain, Lang., and Music, McGill Univ., Rabinovitch House, 3640 rue de la Montagne, Montreal, QC H3G 2A8, Canada, mickael.deroche@mcgill.ca), Aditya M. Kulkarni, Julie A. Christensen (Auditory Prostheses and Percept. Lab., Boys Town National Res. Hospital, Omaha, NE), Charles J. Limb (Dept. of Otolaryngol. – Head and Neck Surgery, Univ. of California San Francisco School of Medicine, San Francisco, CA), and Monita Chatterjee (Auditory Prostheses and Percept. Lab., Boys Town National Res. Hospital, Omaha, NE)

Sensitivity to static changes in pitch has been shown to be poorer in school-aged children wearing cochlear implants (CIs) than children with normal hearing (NH), but it is unclear whether this is also the case for dynamic changes. Yet, dynamically changing pitch has considerable ecological relevance in terms of natural speech, particularly aspects such as intonation, emotion, or lexical tone information. This study examined children and

adults, with NH or wearing a CI using clinically assigned settings with envelope-based coding strategies. Percent correct was measured in one- or three-interval two-alternative forced choice tasks, for the direction or discrimination of harmonic complexes based on a linearly rising or falling fundamental frequency. Sweep rates were adjusted per subject, in a logarithmic scale, so as to cover the full extent of the psychometric function. Data for up- and down-sweeps were fitted separately, using a maximum-likelihood technique. Hits and false alarms were then converted into d' and beta values, from which a threshold was extracted at a d' of 0.77. Thresholds were very consistent between the two tasks and considerably higher (worse) for CI listeners than for their NH peers. Thresholds were also higher for children than adults. Factors such as age at implantation, age at profound hearing loss, and duration of CI experience did not play any major role in this sensitivity. Sweep direction thresholds held the most predictive power for performance in tasks related to speech prosody.

4aPP8. Neural activation patterns indicate robust auditory nerve synchrony generated with octave-band chirps. Ivy Thompson and Brian Earl (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave, Cincinnati, OH 45267, thompsiv@mail.uc.edu)

Recent research has suggested that the amplitude of high-intensity compound action potentials (CAPs) can detect auditory nerve damage that is missed by traditional threshold measurements. The use of octave-band chirps may enhance neural synchrony and provide clinicians and researchers a precise tool for identifying changes in auditory nerve integrity. A group of Mongolian gerbils ($N=12$) was used to compare the difference in amplitude-intensity functions and neural activation patterns generated by octave-band chirps and tonebursts at octave frequencies between 2 and 16 kHz at 40, 60, and 80 dB SPL. Amplitude-intensity functions revealed larger absolute CAP amplitudes for octave-band chirps than for tonebursts. Neural activation patterns were constructed by plotting the derivative of CAP amplitude across multiple conditions of simultaneous broadband noise that was high passed in 1/3 octave intervals. Neural activation patterns for octave-band chirps and tonebursts showed that the frequency location of peak activation was generally equivalent while the height of peak activation was greater for octave-band chirps than for tonebursts. This suggests that octave-band chirps elicit more synchronous neural firing, thereby indicating that they could be an optimal stimulus for detecting regional damage in auditory neurons that encode supra-threshold stimuli.

4aPP9. Effects of elevated amplitude modulation discrimination threshold on simultaneous amplitude modulation rate discrimination. Sean R. Anderson, Alan Kan, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, Office 564, 1500 Highland Ave., Madison, WI 53705, sean.anderson@wisc.edu)

Cochlear-implant (CI) users struggle to understand speech in difficult listening environments partly because they have limited access to auditory cues that allow for perceptual segregation between target and competing sounds. Because CI speech coding strategies optimize the encoding of envelope cues to promote better speech intelligibility, envelope may be an important cue for source segregation in CI users. Recent work has suggested that access to amplitude modulation (AM) depth and rate varies according to location of electrodes in the cochlea, and that turning off electrodes that yield poorer AM sensitivity may improve speech reception. The purpose of this study was to investigate the role of AM sensitivity in the ability of listeners to use AM rate information in multiple electrodes. It was hypothesized that electrodes with higher AM thresholds, which were simulated in normal-hearing listeners, would limit listeners' ability to segregate AM rates. Subjects discriminated between pairs of stimuli that varied in AM rate relative to a reference rate. Stimulus pairing was within- or across-ears. Results suggest that AM insensitive electrodes impair CI users' ability to discriminate AM rates, which may limit source segregation using envelope cues in complex listening environments. [This work was supported by NIH-NIDCD R01 DC003083 (Litovsky).]

4aPP10. Effects of a precursor on amplitude modulation detection are consistent with efferent feedback. Ali Almishaal and Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1217 BEH S, Salt Lake City, UT 84112, ali.almishaal@utah.edu)

The acoustic waveform of speech is characterized by slowly varying amplitude fluctuations (i.e., envelope) and an accurate representation of the envelope is essential for speech understanding. The post-cochlear representation of the contrast between peaks and valleys of the envelope (peak-to-valley contrast) may be reduced by cochlear compression. This study tested (1) whether amplitude modulation (AM) detection thresholds are consistent with cochlear compression and (2) whether the introduction of a precursor before the carrier results in improved AM thresholds, consistent with a decompressed cochlear response via the medial olivocochlear reflex (MOCR). In the no precursor condition, AM thresholds worsen at mid-levels, consistent with reduced peak-to-valley contrast from cochlear compression. In the precursor condition, AM thresholds improved at low modulation frequencies and mid-to-high levels, consistent with a reduction in cochlear amplifier gain and decompression of the cochlear input/output function via the MOCR. These findings suggest the MOCR may play a role in the perception of other amplitude-modulated stimuli, such as speech.

4aPP11. How well do measures of the precedence effect based on clicks predict performance for long-duration stimuli? M. Torben Pastore and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., 4 Irving Pl., Troy, NY 12180, m.torben.pastore@gmail.com)

Despite multiple reflections off nearby surfaces, listeners often localize sound sources based primarily upon the first arriving, direct sound. This is called the precedence effect. Much has been learned about the precedence effect using transient clicks, but the vast majority of everyday sounds are relatively long in duration. Recently, Pastore and Braasch (JASA, 2015) tested the effects of increased lag intensity on localization dominance for longer, 200-ms duration stimuli with 20-ms cosine-squared ramps. Assuming that interactions at the onset of lead-lag stimuli are primarily responsible for the precedence effect, it has been suggested that the binaural cues important to the precedence mechanism are the same for clicks and longer-duration noise stimuli. To test this hypothesis, we presented lead-lag stimuli composed of 1-ms, rectangular clicks, as well as 41-ms and the previously used 200-ms long noise bursts. Five, lead-lag delays between 1 and 5 ms were tested for lead/lag level differences of 0-, –4-, and –8-dB. Results and statistics clearly show that the classic click stimuli and long-duration stimuli do not yield the same performance for the conditions tested in these experiments, especially when lag level is increased. [Research supported by NSF 1320059.]

4aPP12. Backward masking determination with simultaneous early, middle, and late evoked potentials. Silas Smith, Robert Sears, and Al Yonovitz (Dept. of CSD, The Univ. of Montana,Missoula, MT 59812, al.yonovitz@umontana.edu)

Backward masking (BM) functions have been shown to relate to age, lead toxicity, and are differentiated in children with language disorders. These functions may be indicative of auditory processing deficits. This study investigated if evoked potentials (EP) could be utilized to obtain BM functions. A tonal stimulus, followed by an ISI and a noise masker was the EP stimulus. All were studied individually in the appropriate temporal alignment. The design of this study allowed observation of the early, middle, and late auditory evoked potentials. This study randomly presented four different stimulus conditions: (1) tone alone, (2) noise alone, (3) tone and noise, and (4) silence. With a long inter-trial interval (1 s) and high sample rate (25600 Hz) EP's were obtained for 4000 trials. The stimuli were pure-tones (1000 Hz, 10 ms duration with a Blackman function and noise bursts of varying intensity and varied ISI. Results indicated that EP's could be arithmetically combined to observe the differential electrophysiological responses and neurologic loci of evoked potentials during the BM effect.

4aPP13. The inter-stimulus interval critical value in backward masking testing. Robert Sears, Silas Smith, Sarah Schied, and Al Yonovitz (Dept. of CSD, The Univ. of Montana, Missoula, MT 59812, robert.sears@umontana.edu)

Backward masking (BM) has been studied both as a psychoacoustic phenomenon and as a potential diagnostic indicator of auditory processing difficulty. In BM assessment, the subject responds to a brief tonal signal followed by an inter-stimulus interval (ISI) of silence and then by a noise masker. In all previous studies, an adaptive auditory threshold is obtained at each ISI. This study used unique instrumentation that allowed the adaptive ISI to maintain threshold. Twelve subjects were utilized. All had normal hearing with an age range of 18–30 years. The critical ISI was varied with an adaptive procedure in 2 ms steps. The critical ISI value was obtained for stimulus intensity levels above the tonal threshold found in a longer ISI condition. This procedure allowed less fatigue and provided an easier task for the subject compared to the tonal intensity tracking task.

4aPP14. Discrimination thresholds for level increments in the stop-consonant noise bursts of CVC words: Intra-speech masking, temporal-position, temporal-variability, and place-of-articulation effects. Blas Espinoza-Varas (Commun. Sci. & Disord., OU Health Sci. Ctr., 1200 N. Stonewall Ave., Oklahoma City, OK 73117-1215, blas-espinoza-varas@ouhsc.edu) and Jaremiah Hilton (Biostatistics and Epidemiology, OU Health Sci. Ctr., Oklahoma City, OK)

Estimates of sensitivity to level differences in stop-consonant noise bursts could help designing speech-processing strategies such as dynamic-range compression and optimal consonant-vowel intensity ratio. In normal-hearing participants, we measured level-discrimination thresholds (LDTs) for the noise bursts of CVC words (/pæt/, /pæk/, and /kæt/). With /pæt/, a 2I-2AFC task measured LDTs for the pre- or post-vocalic burst in isolation or in word context; in the latter, the burst with level increments (pre or post) remained the same or varied unpredictably from trial to trial. In isolation, the 1.98–2.22 dB LDTs approached those of like-duration random noise, but increased to ≈9.0 dB for the pre-vocalic burst in-context, and more so in unpredictable-burst conditions (10.81 dB); for the post-vocalic burst the LDTs increased only slightly. To assess the role of set-size, within-trial standard, and place-of-articulation, the in-context, unpredictable-burst LDTs were measured with single- or two-observation interval 2AFC tasks presenting only /pæt/ or, randomly, /pæt/, /pæk/, or /kæt/. The context, temporal-position, predictability, and place-of-articulation effects were significant but those of within-trial standard and set size were not; for pre-vocalic bursts, LDTs were much higher (9.97 dB) for the velar than the labial consonant (4.78 dB).

4aPP15. Comparison of Blackman, linear rise-fall, and linear rise-fall chirp signals in backward masking. Robert Sears, Silas Smith, and Al Yonovitz (Dept. of CSD, The Univ. of Montana, Missoula, MT 59812, robert.sears@umontana.edu)

Backward masking is finding applications in clinical testing and the diagnosis of auditory processing disorders. Masking of a tonal signal by a band-limited noise in a backward masking paradigm requires significant vigilance and produces fatigue. This study compared the backward masking functions of ten normal hearing young (18–30 years) subjects using tonal stimuli with different signatures in order to provide a better contrast to the noise masker. Subjects adaptively tracked the tonal threshold using 2 dB changes. The noise stimulus was 70 dB HL, and the inter-stimulus interval (ISI) varied from 2 ms to 64 ms. The results will be described in terms of fatigue, tracking efficiency, and latency.

4aPP16. Backward masking of vowel-consonant stimuli. Allie Cope, Kendra Foster, Robert Sears, Silas Smith, and Al Yonovitz (Dept. of CSD, The Univ. of Montana, Missoula, MT 59812, allie.cope@umontana.edu)

Backward masking (BM) has shown differential effects with age and auditory processing. BM may be related to reduced speech discrimination. Ten normal hearing subjects (18–30 years) were subjects in this study. This study utilized 21 VC stimuli followed by a white noise masker. The ISI interval was 5 ms, and the noise duration was 50, 100, and 200 ms. Each stimulus was randomly presented ten times. Confusion matrices determined consonant intelligibility and information transmission for distinctive features. Thus, the study investigated the accuracy of perception for English consonants as well as determined which of the distinctive features (voicing, nasality, continuancy, sibilancy, frontness, sonorancy, and labiality) remained preserved or changed when exposed to backward masking listening conditions. The results indicated a reduction in these selected features as well as reduced consonant identification as BM became more effective.

4aPP17. Backward masking and tonal audibility before and after a noise burst. Robert Sears, Silas Smith, and Al Yonovitz (Dept. of CSD, The Univ. of Montana, Missoula, MT 59812, robert.sears@umontana.edu)

This study reports on the perceived position of the tonal signal during a threshold tracking procedure. The stimulus was a 1000 Hz sinusoid tone. The inter-stimulus (ISI) interval varied from 2 ms to 64 ms. The noise stimulus was a 70 dB HL band-limited white noise. The subject tracked threshold (2 dB steps). In these experiments, ten normal hearing subjects (18–30 years) were presented the stimulus before the noise burst. Moving through a critical ISI value, the stimulus becomes inaudible after a further reduction in ISI, the stimulus is detected but appears to follow the noise. This finding supports a hypothesis that the tonal signal may be modulated by physiological loci by changing the ISI and the tonal intensity.

4aPP18. Sound recognition depends on real-world sound level. Sam V. Norman-Haignere and Josh H. McDermott (Brain & Cognit. Sci., MIT, 43 Vassar St., Rm. 4141, Cambridge, MA 02139, svnh@mit.edu)

How does the auditory system recognize instances of the same sound class with distinct acoustic properties? As a case study, we investigated the recognition of environmental sounds at different levels. In principle, level-invariant recognition could be achieved by a normalization mechanism that removes variation in level from listeners' representation of sound identity. Alternatively, listeners could use level as a cue to aid their recognition, taking advantage of the fact that different sound classes are typically heard at different levels. The latter hypothesis predicts that sounds heard at atypical levels should be more difficult to recognize. We tested this prediction by asking human listeners to identify 300 environmental sounds, each presented at seven different sound levels between 30 and 90 dB SPL. We grouped these 300 sounds into those typically heard at low (e.g. salt-shaker) and high sound levels (e.g., jackhammer) using ratings collected via Mechanical Turk. For typically loud sounds, recognition accuracy improved monotonically with increasing experiment level. But for typically quiet sounds, recognition accuracy declined at high experiment levels (above 60 dB). These results are consistent with the hypothesis that listeners recognize individual sounds by internalizing the unique distribution of acoustic features that characterize them.

4aPP19. Evaluating whether increment detection at mid-to-high pedestal frequencies is consistent with cochlear compression. Jessica Chen and Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1217 BEH S, Salt Lake City, UT 84112, jessica.chen@utah.edu)

The ability of the auditory system to encode the amplitude fluctuations of a signal is important for processing complex stimuli, such as speech. A study by Florentine [Florentine *et al*., J. Acoust. Soc. Am. **81**, 1528–1541 (1987)] measured intensity discrimination for multiple frequencies as a function of stimulus level. They observed that, for high frequency stimuli, difference limens were poorer at mid levels than at higher or lower levels. This mid-level "hump" is consistent with cochlear compression and suggests that compression may limit the intensity resolution of the auditory system under certain circumstances. To test the generalizability of this interpretation, the current study measured increment detection for conditions associated with the mid-level hump, based on the assumption that increment detection and intensity discrimination are determined by similar physiological processes. This study is part of a larger series of experiments in our laboratory that test whether intensity perception is consistent with peripheral processes such as cochlear compression and efferent-mediated regulation of cochlear amplifier gain.

4aPP20. Effects of interleaved noise on speech recognition in children.

Carla L. Youngdahl (Communicative Sci. and Disord., Saint Mary's College, 45 Madeleva Hall, Notre Dame, IN 46556, cyoungdahl@saintmarys.edu), Sarah E. Yoho, Rachael F. Holt (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), Frederic Apoux (Eye and Ear Inst., The Ohio State Univ. Wexner Medical Ctr., Columbus, OH), and Eric w. Healy (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Normal-hearing adults can isolate frequency regions containing clean speech from surrounding regions containing noise. However, children have been shown to integrate information over a large number of auditory filters, and so they may not be able to isolate frequency regions as well. To assess children's level of auditory filter independence, words were filtered into 30 contiguous 1-ERB-width bands. Speech was presented in every other band, for a total of 15 speech bands. Speech-shaped noise was then added to: all 30 contiguous bands, the 15 bands not containing speech (OFF), or the 15 bands containing speech (ON). Three age groups were tested: 10 adults, 9 older children (6–7 yr old), and 9 younger children (5 yr old). Consistent with previous findings involving consonant recognition, adults displayed large performance differences between off- and on-frequency noise (OFF vs. ON). The 6- to 7-yr-old group performed similarly to adults. In contrast, the 5-yr-old group displayed equivalent performance in the OFF and ON conditions. This indicates that, for these younger children, even noise that was mostly non-overlapping in frequency interfered with speech recognition as much as noise that was on frequency. [Work supported by NIH.]

4aPP21. Development of subcortical pitch representation in three-month-old Chinese infants.

Fuh-Cherng Jeng (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W224, Athens, OH 45701, jeng@ohio.edu), Chia-Der Lin (China Medical Univ. Hospital, Taichung, Taiwan), Grant R. Hollister, John T. Sabol, Garrett N. Mayhugh (Commun. Sci. and Disord., Ohio Univ., Athens, OH), Tang-Chuan Wang, and Ching-Yuan Wang (China Medical Univ. Hospital, Taichung, Taiwan)

This study investigated the development of subcortical pitch processing, as reflected by the scalp-recorded frequency-following response, during early infancy. Thirteen Chinese infants who were born and raised in Mandarin-speaking households were recruited to partake in this study. Through a prospective-longitudinal study design, infants were tested twice: at 1–3 days after birth and at three months of age. A set of four contrastive Mandarin pitch contours were used to elicit frequency-following responses. Frequency Error and Pitch Strength were derived to represent the accuracy and magnitude of the elicited responses. Paired-samples t tests were conducted and demonstrated a significant decrease in frequency error and a significant increase in pitch strength at three months of age compared to 1–3 days after birth. Results indicated the developmental trajectory of subcortical pitch processing during the first three months of life.

4aPP22. Identification of attended speech stream using single-trial electroencephalography recording.

Ala Somarowthu (Dept. of BioEng., Northeastern Univ., Boston, MA), Nai Ding (College of Biomedical Eng. & Instrument Sci., Zhejing Univ., Hangzhou, China), and Ying-Yee Kong (Commun. Sci. & Disord., Northeastern Univ., Dept. of Commun. Sci. & Disord., Northeastern University, Boston, MA 02115, yykong@neu.edu)

Selective attention differentially modulates neural responses to simultaneously presented speech streams. In this study, single-trial EEG classification was performed to identify the attended speech from a two-talker speech mixture. During EEG recordings, normal-hearing listeners paid attention to one speech stream while listening to speech mixtures. The target-to-masker ratios (TMRs) varied from -9 to $+9$ dB. Individual speech streams were processed with head-related transfer functions to simulate different spatial locations. Two simulated spatial conditions (0 vs. $+/-90$ and $+45$ vs. -45 degree azimuth) were tested for each TMR. Features related to (1) cross-correlation values between EEG signals and temporal envelope of each speech stream, or (2) correlation values between reconstructed speech from the EEG signals with the acoustic stimuli, were fed to the classifiers. The dimensionality of the feature vector was reduced using Principal Component Analysis. Linear Discriminant Analysis and Support Vector Machine were used to classify the EEG signals. Classifiers were trained and tested with a five-fold cross validation method on data pooled across TMRs and source locations for trial lengths from 50 s to 10 s. Average classification accuracy was 85% with a 50 s trial length and maintained high at 70% with a reduced trial length of 10 s.

Session 4aSAa

Structural Acoustics and Vibration, Architectural Acoustics, Noise, and ASA Committee on Standards: Building Isolation from Seismic and Groundborne Vibration

James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Hasson M. Tavossi, Cochair
Physics, Astronomy, & Geosciences, Valdosta State University, 2402 Spring Valley Cir, Valdosta, GA 31602

Chair's Introduction—8:30

Invited Papers

8:35

4aSAa1. Vibration isolation of buildings for control of ground borne noise. George P. Wilson (Wilson, Ihrig & Assoc., Inc., 14 Richelle Court, Lafayette, CA 94549-4821, gpwilson2@gmail.com)

Complete vibration isolation has made it possible to locate new buildings on sites with normally unacceptable levels of ground borne noise. The principles are relatively simple but the isolation details are necessarily limited and, to be successful, the isolation bearing pads or springs must have very specific dynamic and structural properties. Since the late 1980s, there have been a number of successful building isolation projects where the intruding ground borne noise from nearby rail facilities and other sources has been reduced to the threshold of hearing or less. The design principles required include massive, high mechanical impedance foundations, very low acoustical impedance isolation bearings, and relatively high stiffness and mass structure for at least two floors above the isolation level. Typical design details and information are presented regarding successful isolation configurations for entire buildings and for box-in-box isolated spaces within the building. Also included are a discussion on isolation bearing materials which have not been successful, as demonstrated by on-site measurements, and a discussion on the mechanical, acoustical, and durability requirements for successful building isolation bearings. An additional item included is a discussion of the differences between building vibration isolation design principles for noise control and building base isolation design for seismic protection.

9:05

4aSAa2. Vibration isolation for concert hall next to busy street. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

A new, world-class performing arts center is currently being developed in a metropolitan center in the United States. The center will include two grand performance theaters. One of these theaters will be located close to a busy surface street. Vibration measurements conducted at the undeveloped site indicated that groundborne noise from street traffic would be audible within the completed theater unless measures were incorporated into the design to reduce vibration transmitted from the street to the interior of the theater. This will be achieved by structurally separating the performance area of the theater from the surrounding structure by supporting the theater on custom designed, resilient bearing pads. This paper discusses the vibration measurements taken, the projections of groundborne noise and the vibration mitigation measures that were incorporated into the structural design of the theater building to reduce groundborne noise to meet the project design criteria for background noise.

9:35

4aSAa3. Particulate damping media and isolation of ground-borne vibrations. Hasson M. Tavossi (Phys., Astronomy, and GeoSci., Valdosta State Univ., 2402, Spring Valley Cir, Valdosta, GA 31602, htavossi@valdosta.edu)

Ground borne vibration can be dissipated by attenuation, absorption, frictional, and viscous damping before being transmitted to the buildings, by means of a suitable particulate media below the structure. We show that the band-pass behavior of particulate media for mechanical vibrations can also result in significant attenuation of vibrations for frequencies on either side of the pass-band. Sample particulate structures are subjected experimentally to the longitudinal and shear waves of different amplitude and frequencies, generated by electro-mechanical shakers vibrating in different directions. The amount of vibration damping by attenuation, resonance absorption, frictional and viscous dissipations are determined experimentally. The pressure wave and shear-waves, generated experimentally by the electro-mechanical shakers, are combined to simulate realistic ground borne vibrations. Damping media consisting of composite materials with different elastic properties are tested as tuned absorbers. The goal is to determine the damping characteristics for most efficient absorption of vibration, of different intensity and frequency. The results can be used to design tuned dampers for specific ground borne vibrations and determine material characteristics for dissipation of ground borne vibrations at large scale.

10:05

4aSAa4. Measurements of the reduction in ground vibration levels from wave barrier trench. John J. LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Where buildings are adjacent to railroad tracks, there are limited practical applications for controlling the transmission of train vibration. In critical listening spaces like auditoria or government buildings, isolation of the building structure is sometimes undertaken. However, this option is not attractive for most commercial or residential development due to the attendant costs and complications. For these properties, one option for reducing levels of ground vibration is to dig a trench between the tracks and the project to act as a wave barrier. Construction of a deep trench presents a wide variety of construction challenges, and the application of this method has been rare within the United States due to the lack of regulations and sensitivity that might demand such an approach. The authors recently had a project where a wave barrier (trench) was applied, which provided the opportunity to observe the construction of the trench and to measure the resulting reduction in ground vibration levels before and after the installation. Various aspects of this process are discussed.

THURSDAY MORNING, 26 MAY 2016 SALON J, 10:50 A.M. TO 11:50 A.M.

Session 4aSAb

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Nuclear-Powered Thermoacoustics

James E. Phillips, Cochair
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Hasson M. Tavossi, Cochair
Physics, Astronomy, & Geosciences, Valdosta State University, 2402 Spring Valley Cir, Valdosta, GA 31602

Invited Papers

10:50

4aSAb1. Design and fabrication of a fission-powered thermoacoustic in-core sensor. Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab, P. O. Box 30, State College, PA 16804, sxg185@psu.edu), Robert W. Smith (Marine & Physical Acoust., Appl. Res. Lab., State College, PA), James A. Smith (Fundamental Fuel Properties, Idaho National Lab., Idaho Falls, ID), and Brenden J. Heidrich (Nuclear Sci. User Facility, Idaho National Lab., Idaho Falls, ID)

A standing-wave thermoacoustic engine with dimensions identical to an ordinary fuel rod was designed to be placed in the core of the Breazeale Nuclear Reactor on Penn State's campus. The heat necessary to produce thermoacoustics oscillations was provided by two 10 mm long by 5 mm diameter, 7.5% enriched, ^{235}U fuel pellets. Those pellets were contained within a stainless-steel finned heat exchanger that was fabricated by additive manufacturing (3-D printing). The (mass-controlled) resonator was suspended in the surrogate fuel rod using two six-armed leaf springs (spiders) that centered the resonator in the "slotted tube" and allowed longitudinal vibrations of the entire resonator that coupled the oscillatory momentum of the gas within the resonator to the surrounding light-water reactor coolant. A 2.0 MPa mixture of 25% argon and 75% helium provided a trade-off between dipole radiation efficiency, resonator length, and low onset temperature differential, to produce a frequency that was high enough to be above the dominant noise produced by coolant and ^{16}N diffusion pumps. These trade-offs were optimized using the Los Alamos DELTAEC software. [Work supported by Idaho National Laboratory and Westinghouse Electric Co. Fabrication and fueling was completed in collaboration with IST-Mirion.]

11:20

4aSAb2. Using the sounds of nuclear power. James A. Smith (Fundamental Fuel Properties, Idaho National Lab., Idaho National Lab., M/S 6188, Idaho Falls, ID 83415, james.smith@inl.gov), Brenden J. Heidrich (Nuclear Sci. User Facility, Idaho National Lab., Idaho Falls, ID), Vivek Agarwal (Human Factors, Controls and Statistics, Idaho National Lab., Idaho Falls, ID), Michael D. Heibel (Global Technol. Development, Westinghouse Electric Co., Pittsburgh, PA), Robert W. Smith (Marine & Physical Acoust., Appl. Res. Lab., State College, PA), and Steven L. Garrett (Grad. Prog. in Acoust., Penn State, State College, PA)

The generation of sound by heat has been documented as an "acoustical curiosity" since a Buddhist monk reported the loud tone generated by a ceremonial rice-cooker in 1568. Over the last four decades, significant progress has been made in understanding "thermoacoustic processes," enabling the design of thermoacoustic engines and refrigerators. We have developed and tested a thermoacoustic engine that exploits the energy-rich conditions in the core of a nuclear reactor. The heat engine is self-powered and can wirelessly transmit the temperature and reactor power by generation of a pure tone which can be detected outside the reactor. We report here the first use of a fission-

powered thermoacoustic engine capable of serving as a performance and safety sensor in the core of a research reactor and present data from two hydrophones in the coolant (far from the core) and an accelerometer attached to a structure outside the reactor. These measurements confirmed that the frequency of the sound produced indicates the reactor's coolant temperature and that the amplitude (above an onset threshold) is related to the reactor's operating power level. These signals can be detected even in the presence of substantial background noise generated by the reactor's fluid pumps. [Work supported by Idaho National Laboratory and Westinghouse Electric Co.]

THURSDAY MORNING, 26 MAY 2016 SALON E, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Non-Native Speech Perception and Production (Poster Session)

Melissa M. Baese-Berk, Cochair
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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 authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aSC1. Perception of phonological variants by non-native listeners. Hayk Abrahamyan (Psych., SUNY at Buffalo, Univ. at Buffalo, Park Hall 204, Buffalo, NY 14260, hayk@buffalo.edu)

Recent research has demonstrated that listeners process carefully pronounced words (canonical forms) more quickly and accurately than casually produced words (non-canonical, reduced forms), despite the fact that casually produced word forms are more frequent in everyday language use. To date, research on the perception of phonological variants that are typical of casually produced speech has focused, with a few exceptions, on monolingual listeners. The current research examined non-native English speakers' processing of canonical and non-canonical word-forms in an attempt to more fully understand how non-native speakers of English cope with phonological variants in American English. Monolingual American English speakers and non-native American English speakers completed a cross-modal identity priming task with canonical, non-canonical, and unrelated auditory primes and visual targets. Overall, the non-native speakers were significantly slower than native speakers at recognizing both canonical and non-canonical forms, although our data suggest that non-native speakers may encounter more specific difficulties than native speakers when processing phonological variants. Our results constitute an initial attempt at understanding how non-native speakers cope with phonological variation in their second languages.

4aSC2. Children's perception of native dialects and nonnative accents. Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu), Rachael F. Holt, Tiarah Wilcox, Sarah Mabie, and Leah Neczypor (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

In quiet listening conditions, school-aged children can have difficulty understanding nonnative-accented speech whereas adults tend to be highly accurate. The addition of noise substantially depresses word recognition accuracy for both groups. Here, these findings are extended to the perception of an unfamiliar native dialect. Children between the ages of 5 and 7 years (n = 90) were presented with HINT-C sentences produced by three female talkers with different accents—American English (midland dialect), British English, and Japanese-accented English—in quiet or in 8-talker babble with a +4 dB SNR. Results showed highly significant main effects of accent, listening condition (noise, quiet), and age in the expected directions, as well as an interaction between talker accent and listening condition. In quiet, children showed very accurate word recognition for the American and British talkers (97% and 95% correct, respectively) with lower accuracy for the nonnative talker (73% correct). Compared to the quiet condition, performance declined more in the noise-added condition for the British (20% decline) and nonnative talkers (21%) than for the American talker (7%). These results suggest that although school-aged children can understand unfamiliar native dialects, their representations of these dialects may be fragile and highly susceptible to environmental degradation.

4aSC3. The hierarchies of phonetic realization of focus in second language speech. Ying Chen (School of Foreign Studies, Nanjing Univ. of Sci. and Technol., 200 Xiaolingwei St., Nanjing, Jiangsu 210094, China, ychen@njust.edu.cn)

Two experiments were conducted to examine the phonetic realization of focus in L2 Mandarin by L1 American English speakers and in L2 English by L1 Beijing Mandarin speakers. The production data in both experiments indicated an acoustical hierarchy of duration > intensity > F0 at the sentential level. This hierarchy was correlated with another hierarchy in sentence location relative to focus: pre-focus > in-focus > post-focus. In other words, L2 learners tended to produce more nativelike patterns of duration than intensity, and intensity than F0, to code focus. These patterns were more salient in pre-focus condition than in-focus condition, and in in-focus condition than post-focus condition. These findings are consistent with Wu and Chung (2011, ICPhS) and Chen, Xu and Guion-Anderson (2014, Phonetica) that bilingual learners used more duration and intensity than F0 and more in-focus expansion than post-focus compression to code focus in L2 speech. Post-focus compression of F0 was the most difficult acoustic cue in phonetically realizing L2 focus. The nativelikeness of focus realization in L2 speech increased with the increase of L2 experience. [This work was supported by the National Science Foundation of China #61573187 and Fundamental Research Funds for the Central Universities in China NJUSTWGY14001.]

4aSC4. Production of English vowels preceding voiced and voiceless consonants by Korean learners of English. Juyeon Chung (Linguist, Indiana Univ., Memorial Hall 322, Bloomington, IN 47405-7005, chungjulia29@gmail.com)

In English, consonant voicing has large effects on both the quality and duration of the previous vowel, as does the status of the vowel as tense or lax. Our aim is to examine whether there is L1 interference on L2 English vowel productions and whether temporal acquisition is easier than vowel quality acquisition. Korean L2 speakers were chosen as participants; Korean has no contrasts in tense vs. lax vowels, and no coda consonant voicing contrast in a monosyllabic structure, but does exhibit post-vocalic voicing contrasts in disyllabic structures, and in these cases, voiced consonants exhibit lengthening of the preceding vowel. The participants were asked to read the list of English nonce words consisting of the English high vowels and sets of plosives contrasting in voicing as a coda in the two different structures. None of the speakers exhibited different patterns in monosyllabic and disyllabic structures. All speakers did exhibit durational correlates to the voicing contrast, and to the tense-lax distinction. Formant frequency differences were found for both voicing and the tense-lax distinction, however, not as consistently. Specifically, back vowels were not distinguished by a number of subjects. In sum, temporal properties were more easily acquired than vowel quality properties.

4aSC5. Audiovisual integration in fricative production and perception by nonnative speakers. Margaret Harrison, Chao-Yang Lee (Commun. Sci. and Disord., Ohio Univ., Grover Ctr. W252, Athens, OH 45701, mh806711@ohio.edu), and Seth Wiener (Modern Lang., Carnegie Mellon Univ., Pittsburgh, PA)

Lip rounding enhances the acoustic contrast between /s/ and /ʃ/ in English. For speakers of languages without this phonemic distinction, to what degree does lip rounding affect the production and perception of these fricatives? Is visual awareness of lip rounding associated with audiovisual integration commonly observed in the McGurk effect? How does this performance develop as a function of exposure to English? Fourteen learners of English participated in three tasks. First, participants read a list of /s/-/ʃ/ minimal pairs. The spectral center of gravity was measured to examine learners' production of the contrast. Second, participants watched/listened to the same words produced by a native English speaker in four presentation formats: audio only, audiovisual-congruent, audiovisual-incongruent (e.g., an audio /s/ paired with a visual /ʃ/), and visual only. Finally, participants listened to four stop-vowel syllables paired with either congruent or incongruent videos. Two experimental sessions were conducted to examine the development of the performance. Preliminary results showed identification accuracy of /s/, compared to /ʃ/, was affected to a greater extent by presentation format. Response to /s/ was slower in the first session, but was comparable to /ʃ/ in the second session.

4aSC6. International teaching assistants' production of focus intonation. Sophia Kao (Linguist, Stony Brook Univ., SUNY at Stony Brook, Stony Brook, NY 11794-4376, sophia.kao@stonybrook.edu), Jiwon Hwang (Asian & Asian American Studies, Stony Brook Univ., Stony Brook, NY), Hyunah Baek, Chikako Takahashi, and Ellen Broselow (Linguist, Stony Brook Univ., Stony Brook, NY)

We report on a longitudinal investigation of the realization of English focus by 19 Mandarin-speaking International Teaching Assistants (ITAs). Participants read passages containing contrastive information (e.g., *The price of a train ticket is twenty dollars, while the price of a bus ticket is eleven dollars*), and then responded to the experimenter's questions (*Is the price of a bus ticket twenty dollars*?). ITAs were tested within a month of their arrival in the US, and again at the end of their first semester. Overall, the productions of ITAs at both points in time were judged as less natural by native English listeners than the productions of the native speakers of English, though the naturalness of some ITA productions improved at the second sampling. Acoustic analyses of the ITA productions and comparison with the productions of 18 native English speakers revealed a good deal of interspeaker variability in the ITA productions, with several different patterns associated with the "unnatural" productions: (a) failure to accent the focused element; (b) failure to deaccent the word following the focused element; and (c) failure to align the accent with the stressed syllable of the focused word, with the entire focused word spoken on high pitch.

4aSC7. Use of language-specific speech cues in highly proficient second-language listening. Anne Cutler (Univ. of Western Sydney, 1 Tewkesbury Ave., Apt. 60, Darlinghurst, NSW 2010, Australia, a.cutler@uws.edu.au), Laurence Bruggeman (Univ. of Western Sydney, Penrith, NSW, Australia), and Anita Wagner (UMCG, Groningen, Netherlands)

Language-specificity in listening to speech occurs at all processing levels and even between structurally close languages (e.g., English, Dutch). Transitional cues to fricative place of articulation are used in English for identifying /f/ (which resembles theta) but not /s/, whereas in Dutch (without theta) they are used for neither. In spoken-word recognition, suprasegmental cues are used in Dutch, but not in English (with more segmental reduction); Dutch L2 listeners even outperform native L1 listeners in detecting origin of differently stressed English syllables (e.g., car- from CARton versus carTOON). Here, longterm residents in Australia with Dutch as L1 but predominantly using English completed each of these tasks. In the phonetic task, with cross-spliced nonsense words, these listeners performed just as Dutch listeners in the Netherlands, showing insensitivity to transitional cues for both /f/ and /s/. In the lexical task, with word fragments (e.g., car-), they however did not behave as L1 Dutch and outperform Australian English listeners, but instead resembled the latter, by ignoring suprasegmental stress cues. A (lexical) listening strategy available in L1 can apparently be abandoned if it delivers little payoff in L2, but acquiring for L2 listening a (phonetic) strategy not used in L1 seems less feasible.

4aSC8. The perceptual assimilation model for suprasegmentals and cross-language lexical-tone identification. Jennifer Alexander (Dept. of Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, Jennifer_Alexander@northwestern.edu) and Dr. Yue Wang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

We examine how native lexical-tone experience influences identification of novel tone. Cantonese, Thai, Mandarin, and Yoruba listeners identified CV syllables bearing the six phonemic Cantonese tones. Accuracy scores were submitted to a two-way rANOVA with L1-Group (x4) as the between-subjects factor and Tone (x6) as the within-subjects factor. Tone error patterns were also assessed via rANOVAs with L1-Group (x4) as the between-subjects factor and Response-Pattern (% correct versus % other response) as the within-subjects factor. Consistent with previous reports, native listeners' confusions reflected effects of ongoing tonal mergers and a crowded tone space. Non-native listeners appeared to assimilate novel tones to L1 tone categories by attending to phonetic cues relevant to the phonological and phonetic properties of their L1s. Overall, results support predictions of the Perceptual Assimilation Model for Suprasegmentals (PAM-S). [Support: NSF grant 0965227 to J.A.A.]

4aSC9. The language-familiarity effect in talker identification by highly proficient bilinguals depends on second-language immersion. Sara C. Dougherty and Tyler K. Perrachione (Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, sarad12@bu.edu)

Listeners identify voices more accurately in their native language than an unfamiliar, foreign language in a phenomenon called thelanguage-familiarity effect (LFE). Talker identification studies consistently find that even highly proficient speakers of a foreign language persist in identifying voices more accurately in their native language than their second language. It remains unclear why some bilinguals continue to exhibit the LFE despite fluent receptive and expressive second-language skills. In the present study, native Mandarin speakers who were fluent in English and living in the United States learned to identify Mandarin- and English-speaking talkers by the sound of their voice. We assessed hypotheses proposing that either depth of second-language immersion or age of second-language acquisition account for the magnitude of the LFE among highly proficient second-language speakers. Only the extent to which bilinguals were currently immersed in second-language use affected the magnitude of the LFE; native Mandarin speakers who currently used predominately English in their daily lives exhibited no LFE, whereas those who currently used predominately Mandarin continued to have superior native-language talker identification abilities. These results suggest that the magnitude of the LFE in talker identification by bilinguals is a function of real-world immersion in foreign language use.

4aSC10. Vowel length and perception of English as a Lingua Franca. Mara Haslam and Elisabeth Zetterholm (Dept. of Lang. Education, Stockholm Univ., Stockholms Universitet, Institutionen för Språkdidaktik, Stockholm 10691, Sweden, mara.haslam@isd.su.se)

English is one of the most widely used languages in the world but, in contrast to many other languages, a majority of users of English are non-native speakers, meaning that many interactions in English fall under the category of English as a Lingua Franca (ELF). Understanding of how non-native speakers' perception works in English is important for both researchers and English teachers. Jenkins' (2000) Lingua Franca Core (LFC), a list of pronunciation characteristics which are claimed necessary for accurate ELF perception, has already been adopted as a standard for training English teachers and for students of English. However, the research on which the LFC is based is rather limited. An earlier study by the presenters about the importance of the fortis-lenis distinction on stop consonants for ELF perception indicates that participants' perception of word-initial stops cannot be explained easily in relation to voice onset time. This study represents a further attempt to investigate the claims of the LFC using more controlled methods, focusing on the claim that length is important for accurate vowel perception. Word tokens from conversations in English between speakers with different L1s are used. Results of vowel-length measurements and their relationship to perceptual results will be presented.

4aSC11. Visual and auditory native language interference in perceptual learning of non-native speech sounds. Pamela Fuhrmeister, F. Sayako Earle (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., U-1085, Storrs, CT 06269, pamela.fuhrmeister@uconn.edu), Jay Rueckl (Psychol. Sci., Univ. of Connecticut, Storrs, CT), and Emily Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, Storrs, CT)

Learning speech sounds in a second language is challenging for adults, especially when non-native phonemes are perceptually similar to those in the native language. Results from Earle and Myers (2015) suggest that auditory exposure to native language tokens that are perceptually similar to a learned non-native phonetic contrast attenuate sleep-mediated gains in perceptual learning. The present study seeks to determine whether activating abstract phonetic category representations through visual input will produce a similar interference effect on a learned non-native phonetic contrast. To test this, we trained participants to identify the Hindi dental and retroflex contrast and reassessed their performance following a period of sleep. Immediately after training, participants were exposed to interference tokens through a pseudohomophone judgment task, in which they were asked to decide if a string of letters spelled out a real word when read aloud (e.g., "drane" = drain). One group read words with /d/-initial tokens (perceptually similar to the learned contrast) and another read /b/-initial tokens (perceptually dissimilar). The effect of visual interference on perceptual performance differed from the effects observed with auditory interference. These results suggest that different modalities of language input differentially interfere with perceptual learning of non-native speech sounds.

4aSC12. The effect of visual information on non-native speakers' perception of Cantonese tones. Yan Chen (Linguist Dept., Univ. of Arizona, TUCSON, AZ 85721, yanchen@email.arizona.edu)

This study examines the effect of tone marks on the perception of 5 difficult Cantonese tone pairs that have high perceptual similarity: high-rising (25) versus mid-rising (23), mid-level (33) versus low-level (22/11), low-falling (21) versus low-rising (23), low-falling (21) versus low-level (22/11), and low-rising (23) versus low-level(22/11). Native speakers of American English and native speakers of Mandarin participated in a categorial AXB pre-test, a categorial AX training, and a categorial AXB post-test. Half of the subjects received iconic symbols (˨˥ for high-rising (25), ˧ for mid-level (33), ˨˩ for low-falling (21), ˨˧ for low-rising (23), and ˨ for low-level (22/11)) as feedback in the training (Auditory-Visual training group) while the other half did not (Auditory). Preliminary results showed that both language groups improved their discrimination of all the tone pairs and that Mandarin speakers received higher percent response than English speakers in the post-test. AV subjects and A subjects, regardless of language groups, did not differ significantly in percent correct response in the post-test. Reaction time data suggested that AV subjects responded faster on the pair high-rising (25) versus mid-rising (23) than A subjects, regardless of language groups, and that Mandarin speakers responded faster than English speakers in the post-test.

4aSC13. Intelligibility, fluency, and variability in non-native speech. Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Oyer Ctr. B-7, East Lansing, MI 48824, mbaesebe@uoregon.edu), Tuuli H. Morrill (George Mason Univ., Fairfax, VA), and Ann R. Bradlow (Northwestern Univ., Evanston, IL)

Native and non-native speech differ in many ways, including overall speech rate, which tends to be substantially slower for non-native speakers (Guion *et al.*, 2000). Recent work has suggested that non-native speech may be not only slower, but also more variable when non-natives are reading aloud (Baese-Berk and Morrill, 2015). Speaking rate also influences how listeners perceive non-native speech—slower readers are perceived as more accented and less comprehensible (Munro and Derwing, 1998). In the present study, we ask whether variability in speaking rate also has this effect on listeners. Specifically, we ask whether speaking rate variability is correlated with non-native speaker intelligibility and/or with judgments of fluency. In the present study, we ask listeners to transcribe sentences from speakers who show more or less variability in speaking rate across sentences in a paragraph-length reading passage. We also ask them to rate the fluency of the read sentences. Speech samples were taken from native Mandarin, Korean and English speakers' recordings of the English "North Wind and the Sun" passage in the ALLSSTAR collection of digital speech recordings (Bradlow *et al.*, 2010). These results provide insight into the relationships between variability, intelligibility, and fluency in both native and non-native speech.

4aSC14. Variability and stability in native and non-native Japanese speech. Charlotte Vaughn, Melissa Baese-Berk (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403-1290, cvaughn@uoregon.edu), and Kaori Idemaru (East Asian Lang. and Literatures, Univ. of Oregon, Eugene, OR)

The speech of second language learners is often characterized as more variable than the speech of native speakers (e.g., Baese-Berk and Morrill, 2015; Jongman and Wade, 2007; Jongman *et al.*, 2007; Rogers*et al.*, 2012). However, among the relatively few studies which have examined this question directly, the majority have focused on: learners of English, learners from one L1 background, and have examined one linguistic feature per study, leaving open the possibility that non-native speech may not be more variable than native speech in all cases. The present work examines the production of Japanese sentences by native Japanese speakers, and learners of Japanese from two language backgrounds (English and Chinese). Counter to previous results, we find several cases where non-native speakers are *less* variable than native speakers, including on various measures of vowel spectrum variability, and voiced stop realization. Thus, the relationship between variability and language background appears to be more complex than previously thought. Accurately characterizing variability across native and non-native speech has implications for models of second language acquisition, and speech perception. We suggest that future work on variability in non-native speech should examine factors such as speakers' L1 and L2, and multiple linguistic features.

4aSC15. Discriminability of non-native tonal contours in low-pass filtered speech. Tuuli Morrill and Zhiyan Gao (George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, tmorrill@gmu.edu)

Native speakers of a language are able to easily detect non-native "accents"—usually a perceived accent is attributed to phonemic- and phonetic- level information. However, a growing body of work suggests that non-native speakers also exhibit differences from native speakers at the prosodic level. Differences are exhibited in the placement of lexical stress and phrasal accents, as well as the acoustic cues used in prominence production; prosodic differences in production appear to pattern in (native)-language-specific ways (Morrill, *ICPhS Proceedings,* 2015). In this experiment, we ask (1) whether differences in the realization of tonal contours are perceived by listeners, and (2) whether perceptual discriminability is patterning according to speakers' native language. Participants listened to low-pass filtered phrases from the Stella passage recordings on the GMU Speech Accent Archive—produced by native speakers of English, Mandarin, Korean, Arabic, and Turkish—and judged whether speakers were saying the same thing. Results indicate that participants were less likely to rate two phrases as the same (even when they were) if they had been produced by speakers of different native languages. Patterns of discriminability across language pairings are attributed to differences in the number and timing of tonal events (e.g., pitch accents).

4aSC16. Recognition memory for foreign- and native-accented sentences. Kristin Van Engen (Washington Univ. in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899, kvanengen@wustl.edu) and Jonathan Peelle (Washington Univ. in St. Louis, St. Louis, MO)

The acoustic-phonetic properties of speech signals affect not only their intelligibility, but also how well they are encoded in memory. Recognition memory can be improved, for example, when speakers intentionally speak clearly (Van Engen *et al*., 2012). One explanation for this result is that enhanced acoustic-phonetic cues reduce the cognitive effort associated with perceptual speech processing, thereby increasing the availability of processing resources for encoding speech content in memory. In the current study, we tested the hypothesis that, following the same logic, recognition memory would be reduced for foreign-accented speech, in which acoustic-phonetic cues deviate from native-language norms. Participants heard English sentences produced by a native speaker of English and a native speaker of Korean. Recognition memory was tested using an old-new judgment task in which participants heard those recordings again, along with an equal number of new items. Surprisingly, recognition memory was higher for Korean-accented English than for native-accented English. A follow-up study in which test sentences were presented visually showed no difference between foreign- and native-accented speech. These results are consistent with an acoustic distinctiveness account in which foreign-accented speech can in some cases be remembered more accurately due to its distinct acoustic features.

4aSC17. Categorization training for non-native accented word recognition. Rachel Tessmer (Commun. Sci. and Disord., The Univ. of Texas at Austin, 2504 Whitis Ave., A1100, Austin, TX 78712, tessmer.rachel@gmail.com), Eriko Atagi (Volen National Ctr. for Complex Systems, Brandeis Univ., Waltham, MA), Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN), and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

Non-native accented speech is considered an adverse listening condition for native speakers (Mattys *et al*., 2012). Previously, accurate categorization of non-native accents was found to relate to accented speech-in-noise recognition (Atagi and Bent, 2015). The goal of the current study is to examine (1) the extent to which native speakers can be *trained* to categorize the language backgrounds of non-native talkers using feedback, and (2) the extent to which accent categorization training, relative to word and sex identification tasks, improves non-native accented word recognition. Participants were randomly assigned to one of three training tasks: an accent categorization task with trial-by-trial feedback ($n = 10$), a transcription task ($n = 10$), and a speaker sex categorization task ($n = 10$). Critically, all participants were exposed to the same set of training stimuli. Participants completed a transcription task with speech-shaped noise prior to and following training. We assessed keyword accuracy for non-native accented speech and native accented speech. Our results demonstrate that single-session training can improve accent categorization. Accent identification training, word identification, and sex identification all yielded more accurate non-native speech perception (post>pre). We find that irrespective of training task, exposure to non-native accents enhances intelligibility of non-native accented speech.

4aSC18. The bilingual advantage in phonetic/phonological learning: A study of bilingual and monolingual patterns in the reproduction of word-final stops in three novel accents of English. Laura Spinu (Anthropology, Univ. of Western ON, Western University, London, ON N6A 3K7, Canada, lspinu@uwo.ca), Jiwon Hwang (Asian & Asian American Studies, Univ. of Stony Brook, Stony Brook, NY), and Renata Lohmann (Univ. of Western ON, London, ON, Canada)

As part of a larger study investigating the acoustic correlates of accentedness in the reproduction of various accents of English by English monolinguals and French-English bilinguals, we explored speakers' ability to imitate and spontaneously reproduce patterns of realization of word-final coronal stops in three different accents: SE England (Sussex) in which these stops are 100% glottalized, and Russian English and South African English, in which these stops exhibit canonical release about 50% of the time. We have so far fully analyzed the Sussex results and partially analyzed the Russian results. The two groups were characterized by different behaviors: while the bilinguals successfully reproduced the Sussex English accent, the monolinguals did not. By contrast, neither of the groups was successful in reproducing the Russian English accent. After considering the characteristics of each group of speakers and each accent, we conclude tentatively that it is the bilinguals, as a group, who were more successful in the phonetic/phonological learning of a new pattern, perhaps as a result of some type of "bilingual advantage." Based on work by Calabrese (2011) and Krizman *et al*. (2012), we propose that this advantage stems from longer availability of acoustic information in echoic memory.

4aSC19. Transitioning learning strategies in speech categorization enhances lexical learning. Han-Gyol Yi, Rachel Tessmer, and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, 2504 Whitis Ave., A1100, Austin, TX 78712, tessmer.rachel@gmail.com)

A recent model suggests two strategies are involved in speech category learning: a reflective strategy that maps sounds onto categorical representations explicitly, and a reflexive strategy that does so implicitly. Successful learners transition from an initial reflective strategy to a reflexive strategy. We examined the extent to which speech category learning in adults can be improved by selectively enhancing these strategies in an optimal sequence, and how this influences performance in a lexical learning task. Monolingual English speakers ($N = 40$) learned to categorize Mandarin tones. Participants in the "optimal" condition ($n = 10$) were first presented with a combination of feedback information and talker presentation designed to enhance reflective strategies, and later presented with a feedback-talker combination that targeted reflexive strategies. This order was reversed in the "sub-optimal" condition ($n = 10$). Two control conditions ($n = 10$ each) exclusively targeted either strategy. All participants then learned 24 pseudo-Mandarin lexical items across three days. Participants in the optimal condition outperformed those in the sub-optimal and control conditions on the lexical learning task despite the fact that they did not attain the highest speech categorization accuracy. These results suggest that transitioning from reflective to reflexive strategies is ideal for applying learned speech categories to novel words.

4aSC20. Non-native speakers' acoustic variability in producing American English tense and lax vowels. Bruce L. Smith (Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Rm. 1201, Salt Lake City, UT 84112, bruce.smith@hsc.utah.edu) and Rachel Hayes-Harb (Linguist, Univ. of Utah, Salt Lake City, UT)

The primary goal of the present study was to compare native (L1) and non-native (L2) speakers' vowel productions to determine whether the L2 subjects showed similar or greater within-subject (i.e., token-to-token) variability relative to the L1 talkers when producing American English vowels. We were particularly interested in whether L2 subjects whose vowels were

more native-like would be less variable in their productions and whether L2 talkers' whose vowels were less native-like would be more variable in their productions. First and second formants of the tense and lax vowel pairs / i—ɪ /, / e—ɛ /, and / u—ʊ / were measured and Coefficient of Variation was calculated for 10 native speakers of American English and 30 non-native speakers of English from three different language backgrounds (viz., 10 Mandarin speakers, 10 Korean speakers, and 10 Spanish speakers). Overall, the L2 subjects' vowel formant patterns were found to be comparable to those of L1 speakers approximately half of the time. Whether the L2 talkers' American English vowel formants were native-like or not, however, they seldom showed greater within-subject variability than the native speakers.

4aSC21. The effect of speaking style adaptations on speech perception in noise by native and non-native listeners. Kirsten Meemann (English, Univ. of Texas at Austin, 204 W. 21st St., Mail Code B5000, Austin, TX 78712-1164, kirsten.meemann@utexas.edu) and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

It is well established that non-native listeners perform worse on speech perception in noise tasks compared to native listeners. However, few studies have directly compared the effect of speaking style adaptations on speech perception in energetic and informational maskers for these two listener groups. The present study examined intelligibility of conversational (CO), clear (CS) and noise-adapted (NAS) meaningful sentences mixed with speech shaped noise (SSN) and 2-talker (2T) babble. Sentences were presented at −5 dB signal to noise ratio (SNR) to native and −3 dB SNR to non-native listeners. The results revealed that CS and NAS significantly improved word recognition in noise for both listener groups. However, the gains in intelligibility were substantially greater for native compared to non-native listeners. Word recognition was overall better in 2T babble than in SSN. Our findings confirm that both native and non-native listeners benefited from speaking style adjustments, but native listeners were better able to utilize the intelligibility-enhancing modifications. The results also suggest that informational masking in 2T babble is less disruptive than energetic masking in SSN. This may in part be due to the "glimpsing" windows through which the target speech could become more easily accessible.

4aSC22. Adaptation to accent is proportionate to the prevalence of accented speech. Matt Lehet and Lori L. Holt (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Carnegie Mellon, Baker Hall-346, Pittsburgh, PA 15213, mil@andrew.cmu.edu)

Listeners rapidly reweight the mapping of acoustic cues to speech categories in response to abrupt introductions of accented speech. For instance, when encountering an accent that reverses the typical correlation of acoustic cues to speech category membership, listeners rapidly down-weight reliance on secondary cues [Idemaru and Holt, 2011; 2014; Liu and Holt 2015]. Here, we examined the impact of experiencing mixtures of typical and accented speech in evoking cue down-weighting to test how much information the system requires to adapt. Listeners recognized words as *beer* versus *pier* across 41 blocks of 20 trials. In the first block, participants exclusively encountered tokens with the canonical English relationship between fundamental frequency (F0) and voice onset time (VOT); low F0 frequencies were associated with short VOT durations whereas high F0 frequencies were associated with long VOT durations. In subsequent blocks, the ratio of canonical to accented tokens (for which the F0/VOT correlation was reversed) was changed by 5% per block until participants experienced exclusively accented speech. Canonical tokens were then reintroduced incrementally, until participants exclusively heard canonical speech. Reliance on F0 for word recognition linearly decreased as the proportion of accented speech increased. Implications for models of the speech perception system are discussed.

Session 4aSP

Signal Processing in Acoustics and Underwater Acoustics: Detection and Estimation in Uncertain Acoustic Environments I

Paul J. Gendron, Chair
ECE Department, University of Massachusetts Dartmouth, 285 Old Westport Road, North Dartmouth, MA 02747

Chair's Introduction—10:30

Invited Papers

10:35

4aSP1. Information-theoretic analysis of Bayesian localization of a narrowband source in an uncertain environment. Thomas J. Hayward (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Previous work has explored the application of fundamental information-theoretic constructs, including posterior entropy of source location and per-iteration information gain (relative entropy) and their large-ensemble limits, to quantify the performance of iterated (sequential) Bayesian localization of an acoustic source. These information-theoretic quantities are closely tied to Bayesian inference and provide global measures of the uncertainty of source location that is represented by the Bayesian posterior probability density. The present work extends this analysis to the localization of a narrowband acoustic source in an acoustic environment having uncertain sound speed and attenuation represented by a joint pdf of the two quantities. The fundamental principle relating the environmental and acoustic field uncertainties is the change-of-variables theorem of probability theory. The degradation of localization performance due to the environmental uncertainty is quantified by the increase in posterior entropy of source location, interpreted as an information loss. Examples are presented, and extensions to stochastic environments are discussed. [Work supported by ONR.]

10:55

4aSP2. Removing multiple scattering in underwater target detection. Katherine F. Woolfe, Douglas Photiadis, and David Calvo (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC, katherine.woolfe.ctr@nrl.navy.mil)

Recent advances have been made in the field of optics to image through multiple-scattering media in a regime where classical imaging techniques, which rely on the single-scattering approximation, fail. Using results generated by random matrix theory, it is possible to filter the response matrix generated by a source/receiver array to remove the multiple scattering components of the received signals. After removing the multiple scattering components, classical imaging techniques can be used to image with the single-scattering components. We show how this single-scattering filter can be adapted for underwater acoustical imaging in cases where multiple scattering effects are large compared to the reflection from the target. A full-wave 3-dimensional time-domain scaled model is used to characterize filter performance as a function of target strength and environmental coherence length for a shallow-water case. Preliminary results indicate that the filter can be used to detect a target located at least 2 mean free paths away from the source/receiver array.

Contributed Papers

11:15

4aSP3. Additional studies of the acoustics of coffee roasting. Jay R. Johnson and Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, 1 University Station C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu)

Cracking sounds emitted by coffee beans during the roasting process can be recorded by a microphone and used as the basis for automated acoustic roast profiling and monitoring, mimicking what expert artisanal coffee roasters do by ear. Three parameters are used for this purpose. Near the end of the roasting process, sounds known as "first crack" exhibit a higher acoustic amplitude than sounds emitted later, known as "second crack." First crack emits more low frequency energy than second crack. Finally, the rate of cracks appearing in the second crack chorus is higher than the rate in the first crack chorus. This presentation is a companion to the previously published work on the same topic [J. Acoust. Soc. **185**, EL265–EL269 (2014)], but expanded to include a discussion of automated crack detection signal processing, the acoustic characteristics of different coffee beans, and initial results of a study on how individual beans crack and emit their sounds.

11:30

4aSP4. Learning a linear ordered-statistic constant false-alarm rate detector. Timothy C. Havens (Elec. and Comput. Eng., Michigan Technolog. Univ., Houghton, MI), Ian Cummings, Jonathan Botts (Appl. Res. in Acoust. LLC, Culpeper, VA), and Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

The linear ordered statistic (LOS) is a parametrized ordered statistic (OS) that is a weighted average of a rank-ordered sample. LOS operators are useful generalizations of aggregation as they can represent any aggregation, from minimum to maximum, including conventional aggregations, such as mean and median. Hence, we propose an LOS-Constant False-Alarm Rate (CFAR) detector that uses an LOS operator to compute the background level. Specifically, we extend ordered weighted average (OWA) learning methods to learn an LOS-CFAR using two regularization methods, L2 and entropy regularization. We demonstrate our learning process for midfrequency active sonar in cluttered shallow-water environments using both synthetic data and physics-based simulated experiments, and present and discuss the performance of the learned CFAR detectors versus the classical OS-CFAR and cell-average (CA)-CFAR. [Work supported by a NAVSEA Phase I SBIR award.]

11:45

4aSP5. Extrapolation of measured correlation replica fields in passive acoustic source localization. Christopher M. Verlinden (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu), J Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), K G. Sabra (Mech. Eng. Dept., Georgia Inst. of Technol., Atlanta, GA), and W A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

A method of localizing unknown acoustic sources using data derived replicas from ships of opportunity has been previously reported (Verlinden, 2015). The method is capable of localizing sources in positions where data derived replicas are available, such as locations previously transited by ships, tracked using the Automatic Identification System (AIS). Here, we present an extension of this localization method to regions where data derived replicas are not available by extrapolating the measured replicas onto a more extensive grid. This new augmentation provides the additional opportunity for continuous tracking.

THURSDAY MORNING, 26 MAY 2016 — SALON A, 8:00 A.M. TO 11:10 A.M.

Session 4aUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Sediment Characterization Using Direct and Inverse Techniques III

David P. Knobles, Cochair
KSA LLC, PO Box 27200, Austin, TX 78755

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Chair's Introduction—8:00

Invited Papers

8:05

4aUW1. Seabed property inversion—Sequential and direct approaches. Zoi-Heleni Michalopoulou, Tao Lin (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu), and Nattapol Aunsri (School of Information Technol., Mae Fah Luang Univ., Tasud, Muang District, Chiang Rai Province, Thailand)

The goal of this work is to improve our knowledge of the ocean medium using physics of sound propagation and statistical signal processing. Two of the approaches we discuss are based on global and local optimization combined with sequential filtering. A third one falls under the category of direct (“exact”) methods. The first two approaches are based on the extraction and the association between paths/modes and detected arrivals and also produce posterior probability density functions, characterizing the uncertainty structure of modal/multipath arrivals. These are then propagated backwards through an acoustic model for posterior probability density function calculation of environmental parameters. A third method extends Stickler’s exact inverse approach, employing measurements of a reflection coefficient at low frequencies to obtain the sediment sound speed profile. We develop an approximation based on interpolation and a regularization approach. The former improves on results of a previous method for solving the trace equation using a linear approximation. The latter assists in addressing singularities caused by noise in the data. Several assumptions are initially made in order for the approach to work. We show via a sensitivity analysis that the method is robust with respect to initial assumptions and parameter choices needed for both the interpolation and regularization processes. [Work supported by ONR.]

8:25

4aUW2. Modal wavenumber estimation using a Bayesian compressed sensing algorithm. Angelique Dremeau, Julien Bonnel, and Florent Le Courtois (Lab-STICC, ENSTA Bretagne, 2 rue François Verny, Brest cedex 9 29806, France, julien.bonnel@ensta-bretagne.fr)

In shallow water, low-frequency acoustic propagation is described by modal theory. In this context, the environment acts as a dispersive waveguide, and the geoacoustic properties of the seabed can be inferred from the modal wavenumbers. When considering horizontal aperture (using a horizontal array or a towed source), wavenumber estimation is a well-known problem, equivalent to spectral analysis. In this paper, wavenumber estimation is revisited using Compressed Sensing (CS). Our method benefits from two strong physical hypotheses. Only a few modes are propagating so that the wavenumber spectrum is sparse. Moreover, if the source is broadband, the wavenumbers can be related from one frequency to the next using a general dispersion relationship. Our method resorts to a state-of-the-art Bayesian algorithm exploiting a Bernoulli-Gaussian model. The latter, well-suited to the sparse representations, makes possible a natural integration of the prior dispersion information through a wise choice of the Bernoulli parameters. The whole methodology is assessed on simulated data and successfully applied on marine data.

8:45

4aUW3. Maximum entropy estimation of environmental parameter distributions. Richard Pitre (RobTre Res., L.L.C., 58 Crystal Canyon Dr., Carbondale, CO 81623, richard.pitre@robtre.com)

This presentation regards origins and information theoretic foundations of maximum-entropy methods for estimating environmental parameter distributions. Environmental parameter distributions determine uncertainty and confidence levels in transmission loss inputs to mission planning and sonar tactical decision making tools. Given the scale of the oceans and their stochastic-dynamic nature, the sample base for estimation is usually limited relative to the required spatial and temporal resolution. Effective use of estimated environmental distributions for planning and decision making in this context requires well defined and meaningful methods of extrapolative estimation. The principle of maximum-entropy or minimal-presumption provides an important component of such methods. A complete definition of a maximum entropy method requires (1) definition of an application specific information or information base measure that counts the relative number of physically distinct possibilities in measurable subsets of the state space, and (2) a way of imposing minimally informative extrapolation on a combination of data samples, statistical hypotheses, and mathematical presumptions regarding things like smoothness of the underlying distribution. These components of distribution estimation and their interrelationships are discussed.

9:05

4aUW4. Compressive acoustic sound speed profile estimation using wavelets. Michael J. Bianco, Haiqiang Niu, and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr, La Jolla, CA 92037, mbianco@ucsd.edu)

Estimation of ocean acoustic sound speed profiles (SSPs) requires inversion of acoustic data with limited observations. Inversion for true ocean SSP structure is a nonlinear, underdetermined problem that requires regularization to ensure physically realistic solutions. Traditional regularization, which minimizes the energy of best-fit solutions, requires undersampling of true SSPs or using few shape functions. This gives low resolution SSP estimates which can affect the accuracy of other parameters in geoacoustic inversion. Compressive sensing (CS) reliably estimates signal parameters for certain highly underdetermined linear problems provided the signal can be "compressed:" represented in a sparse domain where few non-zero coefficients (out of many) explain the observations. Here, it is shown that ocean SSPs can be compressed using dictionaries of wavelets, empirical orthogonal functions, and other shape functions given *a priori* environmental statistics. Shape dictionaries are sought which represent SSPs using the fewest coefficients. These optimal dictionaries are used with CS to estimate SSPs by inverting a linearized normal mode model. It is shown that range-independent ocean SSPs are estimated robustly and in high resolution using CS.

Contributed Papers

9:25

4aUW5. Representation of depth-dependent gradients in sediment geoacoustics by Bernstein polynomials. Jorge E. Quijano, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. (Ring Road), Victoria, BC V8P 5C2, Canada, sdosso@uvic.ca), Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA), and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

We present a seabed parametrization approach for depth-dependent gradients in sediment geoacoustics, a property commonly observed in muds. The method represents continuous functions by a polynomial form, consisting of a finite sum of Bernstein basis weighted by real coefficients which are estimated by Bayesian geoacoustic inversion of seabed reflectivity data. The advantages of the Bernstein representation of continuous gradients are discussed, including efficiency in representing a wide variety of gradients with only a few coefficients, as well as high numerical stability of the polynomial form to perturbation of its coefficients. The performance of the Bernstein parametrization applied to geoacoustic inversion is illustrated with simulated data obtained from a realistic seabed scenario. In addition, the Bernstein approach is applied to experimental data from four mud sites at the Malta Plateau. The estimated geoacoustic profiles are in good agreement to core measurements from the area, and serve to illustrate the ability of the Bernstein-based inversion to represent steep gradients. Comparison to results obtained by discrete (multi-layered) and other continuous gradient representations (line-, sinusoid-, and spline-based) is presented.

9:40

4aUW6. On connection between statistical integral equations and statistical inference in ocean acoustics and sonar. David P. Knobles (Knobles Sci. and Anal., LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com) and Preston S. Wilson (ARL, Univ. of Texas, Austin, TX)

Under what circumstances do statistics of source and waveguide parameter values derived from an application of statistical inference, such as Bayesian or Maximum Entropy, reflect the temporal and spatial inhomogeneities of the waveguide? The statistics of the objective function is influenced by model error that reflects the non-deterministic features of the

waveguide and the source. This connectivity is explored with a new derivation of statistical equations previously discussed by Dozier and Tappert. The resulting statistical integral equations provide the relationship between the fluctuations in the waveguide to those of the acoustic field. Then, the idea is to infer, from an ensemble of measured acoustic data, marginal probability distributions for the Green's functions and mode coupling matrix coefficients. This is accomplished by inferring the statistics of the source and the physical properties of an ocean waveguide such as range-dependent sound speed layers in the water column and the seabed, and layer interface roughness parameters. The inferred statistics of the mode coupling operator and Green's functions, and the noise spatial correlation then become input into the statistical integral equations. In principle these modeled acoustic field statistics can then be compared to the measured field statistics. [Work supported by ONR Code 32 OA.]

9:55–10:10 Break

10:10

4aUW7. A new time-domain model for bottom backscatter. Dajun Tang and Darrell R. Jackson (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Bottom backscatter is important for a number of underwater applications: it is a source of noise in target detection and a source of information for sediment classification and geoacoustic inversion. Bottom backscatter measures sound scattered from roughness and volume heterogeneity in different sediment layers. While current models can successfully handle scattering from such sediment layers, they cannot correctly predict the scattered intensity as a function of time. A new model for bottom backscatter is introduced which provides an exact solution for scattered intensity as a function of time under first order perturbation theory. Examples of backscatter from various sediment layers will be shown in contrast to previous models. Issues such as the "spherical wave effect" will be addressed, and applications to model shallow water reverberation will also be discussed.

10:25

4aUW8. Spatial variability of seabed properties on the Malta Plateau inferred with an autonomous underwater vehicle. Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3P6, Canada, jand@uvic.ca), Charles W. Holland (Appl. Res. Lab., Penn State Univ., State College, PA), Stan E. Dosso, Jorge E. Quijano, and Eric Mandolesi (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

We develop an automated method to infer geoacoustic properties along tracks surveyed by autonomous underwater vehicles (AUVs). The AUV tows a 32-element receiver array and a source emitting signals ("pings") at regular intervals. Recordings are processed in terms of reflection coefficients, resulting in large data volumes with substantively more information on seabed structure than traditional seismic profiling. However, interpreting seabed spatial variability requires efficient inversion. The inverse problem is non-linear and requires Bayesian sampling to quantify parameter uncertainties. To account for changes in the number of seabed layers at each ping position, the parametrization treats this number as unknown with a Poisson prior and even-numbered order statistics to improve efficiency. The method is applied to 340 data sets along a 14-km track on the Malta Plateau, employing 8 graphics processing units for approximately 2 weeks of computing time. The results resolve layering along the track with previously unreported detail. An erosional boundary is clearly resolved as a high-velocity, high-density layer and appears rougher and is buried deeper in shallower water. Depressions along this boundary are filled in with lower-velocity material. In addition, sound attenuation is well constrained in a thick low-velocity wedge. [Work supported by ONR and SERDP.]

10:40

4aUW9. Estimation of geoacoustic model parameters from modal amplitude information. N. Ross Chapman (School Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada, chapman@uvic.ca) and Rui Duan (Inst. of Acoust. Eng., Northwestern Poly-Tech. Univ., Xi'an, China)

This paper analyzes an approach for estimating geoacoustic model parameters from information contained in normal modes of a broadband signal. Propagating modes are resolved by time-warping deconvolved signals from light bulb sound sources deployed at short ranges in shallow water. Amplitudes of the resolved modes contain information about sediment sound attenuation through the modal attenuation coefficient. However, the coefficient also depends on the sediment sound speed and density. A sequential inversion approach was developed that enables effective use of modal amplitudes to estimate sound attenuation in sediments. The inversion is based on a sequential Bayesian approach applied to features of resolved modal data that are highly sensitive to specific geoacoustic model parameters. Travel times of modal frequency components are inverted first for sediment sound speed and sediment layer thickness, and these estimates are used in subsequent stages. The effects of errors in estimates from previous stages are analyzed for the impact on estimates of sound attenuation in the final stage. In particular, it is shown that the sediment density is weakly sensitive and does not have significant impact on the estimation of attenuation.

10:55

4aUW10. Effects of seabed curvature on the scattered acoustic field. Sheri Martinelli and Charles Holland (Marine & Physical Acoust., ARL Penn State, P.O. Box 30, Mailstop 3230D, State College, PA 16804-0030, slm77@psu.edu)

Knowledge of the acoustic properties of the sea bottom provides valuable input into the development and performance of information processing algorithms designed to detect and identify, e.g., man-made objects near to, or embedded in, the seabed. This information is also useful for improving modeling and simulation results in reverberation-limited environments. A critical challenge in this area is in allowing for range dependent bathymetry as it can affect the propagating acoustic field in ways that are difficult to predict. Typical approaches to modeling variable bathymetry assume either piecewise flat or tilted (linear) geometry; however, this only serves as a discrete approximation to reality. This study investigates the effects of mild curvature on the 3D acoustic field in frequency and angle space via finite difference time domain simulations. We present results and discuss the implications on forward modeling of the acoustic interaction with the seabed. [Work supported by the Strategic Environmental Research and Development Program (SERDP).]

Session 4pAA

Architectural Acoustics and Musical Acoustics: Opera Rehearsal and Performance Spaces

Robin S. Glosemeyer Petrone, Chair
Threshold Acoustics.com, 53 W Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—1:00

Invited Papers

1:05

4pAA1. Looking to history for a modern paradigm in opera house design. Robin Glosemeyer Petrone and Scott D. Pfeiffer (Threshold Acoust. LLC, 141 West Jackson Blvd., Ste. 2080, Chicago, IL 60604, robin@thresholdacoustics.com)

The design of a 600-seat opera house is a rare opportunity. The cost of modern opera production leads to much larger facilities and small venues are more frequently multi-purpose venues by nature, leading to the use of concert hall or lyric theatre forms that can be pressed into service for opera. The United States has many small theatres, mostly on college campuses, but few see frequent use for Opera and as a result design choices tilt away from optimal for Opera performance. Historic opera houses of Europe present more opportunities for exploration. Many follow the Italianate form which is intimate and evokes our sense of what an opera house "should" be, though closer investigation exposes the weakness in these designs both theatrically and acoustically. The experience for many patrons is removed and lonely with severely challenged sightlines and an acoustic experience that is frontal, dry, and unengaging. Our exploration took us to the Royal Opera in the Palace of Versailles whose form varies in key ways from the typical. While the construction techniques of Versailles hinder its acoustic response, as inspiration, a new form for intimate opera is created.

1:35

4pAA2. Navigating the design of a 600-seat opera house. Scott D. Pfeiffer and Robin Glosemeyer Petrone (Threshold Acoust. LLC, 53 West Jackson Blvd., Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

Negotiation of a design for a 600-seat opera and chamber music hall presents challenges of dimension. Can the main floor audience enter under the parterre? What is the relationship of the singer to the closest audience members, leaving room for a full-size orchestra? Two full concept designs were carried to fully plausible outcomes to test these questions as well as the question of the resulting acoustic environment in order to arrive at decisions amongst the design team and user groups. The power of auralization and visualization tools was brought to bear on the decision making process, supplementing comprehensive benchmarking studies. While each constituent went into the process with moderately to significant biases, all entered the final discussions having arrived with an open mind about the optimal outcome for the unique circumstances of the project. Communication among the team using the medium of greatest familiarity for multiple constituents led to an outcome that has broad support from the entirety of the design team.

2:05

4pAA3. The golden throat—Effect of proscenium zone shaping in opera houses on balance between stage and pit. Joseph W. Myers (Kirkegaard Assoc., 801 W. Adams St. 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

This paper investigates the importance of the walls and ceiling in the "throat" of an opera house—the zone surrounding the proscenium—for encouraging a balance that favors onstage singers over musicians in the orchestra pit. By looking at a variety of opera houses, the author tests the idea that surfaces in this zone that are solid and not strongly angled are especially beneficial for encouraging balance.

2:35

4pAA4. Opera acoustics in multi-use performing arts centers. Mark Holden (Jaffe Holden Acoust., 114A Washington St., Norwalk, CT 06854, MHolden@JaffeHolden.com)

There is very little information available on practical solutions for designing acoustics for an orchestra pit. This paper guides the reader through a brief history of orchestra pits and how to design them for the modern multi-use halls used by opera companies. Musicians generally abhor playing in a pit. They feel it is crowded, hot, loud, and a horrible environment in which to hear themselves and other musicians. The acoustician's goal is to improve the acoustic environment within the pit and find the optimal projection of music to the audience and stage. Drawings and details along with photos from pits designed for halls around the world will be presented from Mark Holden's new book Acoustic of Multi-Use Performing Arts Centers published by CRC Press in January.

3:05–3:20 Break

3:20

4pAA5. Improving listening conditions in partially covered opera pits or "The strings are working so hard…why can't I hear them?". Christopher N. Blair and Paul H. Scarbrough (Akustiks, LLC, 93 North Main St., Norwalk, CT 06854, pscarbrough@akustiks.com)

Since the introduction of the gargantuan orchestra to the world of opera by Wagner, opera house designers have been faced with the question of whether to place the orchestra in an open pit, a covered pit, or a partially covered pit. The open pit has long been favored by musicians but has it operational and economical drawbacks, including often large distances from conductor to singer, challenging pit/stage balance conditions, and reduced audience seating close to the stage. A fully covered pit works well for the special works that the Bayreuth master intended for this condition, but is inappropriate for the rest of the core operatic repertoire. The partially covered pit where some upstage musicians play under the stage is conceptually advantageous for a number of reasons, but can present challenges in uneven listening conditions within the orchestra (excessive loudness, ensemble difficulties) and in the house. In some instances, musician complaints have resulted in regulatory action. This paper explores the root causes of ensemble issues commonly found in partially-covered orchestra pits and presents specific solutions drawn from the consulting (and conducting) experience of the authors.

Contributed Papers

3:50

4pAA6. Design of Gran Teatro Nacional in Lima—A world class opera house in South America. José A. Nepomuceno and Julio Gaspar (Acústica & Sônica, Rua Fradique Coutinho, 955 cjt 12, São Paulo, São Paulo 05433-000, Brazil, info@acusticaesonica.com.br)

The design of Gran Teatro Nacional in Lima, Peru, started in 2009 as very special task. Peru needed a venue with world-class standards to host local and international events of opera, concerts, and ballet. The theater has a built area of about 26000 m^2 with 1500 seats capacity. It has a fully equipped fly tower with motorized winches, an 11 m tall acoustical shell, 4 double deck stage lifts, 2 orchestra lifts, and a pit for 100 players. The equipment complies with the most restrictive European standards. The acoustics oriented the design, from room shaping to its volume, from materiality to seats selection. The acoustical consultants assumed modified horseshoe plan to improved intimacy and three tier levels. The hall has two reverberation chambers and several acoustical banners to provide variability in the acoustical response of the hall. Computer modeling and physical model supported the design. Since the Gala Opening in 2012 the theater has been getting remarkable reviews from conductors, international orchestras, and artists in general. The theater complied with the mission of bringing a world class performing art center to Peru. The paper provides the history of the project, architectural, and construction details and acoustical data.

4:05

4pAA7. Improving orchestra and singer communication at the Brazilian Opera. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

Large opera stages present a challenge in maintaining optimum synchronization for principals or a chorus that are far upstage of the orchestra. Further complications arise when the orchestra located in a deep orchestra pit (typical) and only the conductor is visible. This condition induces disparity between the visual timing from the conductor and the arrival of sound from the orchestra. This paper describes how the implementation of electronic architecture solved the problem and improved acoustical conditions on the stage. In addition, the system was extended to the house to provide optimum conditions for symphony performances.

4:20

4pAA8. Analysis and mitigation design of acoustics for performance hall at the Catholic University. Jonathan P. Coyle, Christopher J. Kramer, Nicole A. Bull, Jacob Maclin, and Joseph Vignola (School of Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC, DC 20064, 36coyle@cua.edu)

This presentation will discuss a student level assessment designed with the goal to analyze a Music Hall located on the campus of The Catholic University of America. Ward Recital Hall is 97,712 ft^3, seating 108 and was built as a hall intended for liturgical musical performance, but since the early 1900's the purpose of the room has changed, since then minor improvements have been implemented. We have found that the three significant problems within the room are the HVAC noise contamination, stage projection homogeneity, and sound transmission through barriers (doors and windows). We will present the findings on each of the problems above in how the stage projection is not currently homogeneous, the issue with exterior noise contamination, and how the HVAC can directly affect the listening of viewers. We also will present remedies for each issue as well as the associated cost for the remedies. We will have a list for these problems with several different solutions, with the intent of having a good, better, and best solution with costs with respect to predictions of performance.

Session 4pAO

Acoustical Oceanography and Animal Bioacoustics: Acoustical Oceanographic Tools for the Study of Marine Ecosystems

David R. Barclay, Cochair
Department of Oceanography, Dalhousie University, PO Box 15000, Halifax, NS B3H 4R2, Canada

Wu-Jung Lee, Cochair
Psychological and Brain Sciences, Johns Hopkins University, 3400 N Charles St., Ames Hall 132, Baltimore, MD 21218

Chair's Introduction—1:00

Invited Papers

1:05

4pAO1. Relative impact of signal-to-noise ratio and propagation effects on the performance of an aural classifier. Carolyn Binder (Oceanogr. Dept., Dalhousie Univ., LSC Ocean Wing, 1355 Oxford St., PO Box 15000, Halifax, NS B3H 4R2, Canada, carolyn.binder@dal.ca) and Paul C. Hines (Dept. of Elec. and Comput. Eng, Dalhousie Univ., Halifax, NS, Canada)

Passive acoustic monitoring (PAM) is used to study marine mammals in their habitats, which cover diverse underwater environments. The distinct propagation characteristics of different ocean environments alters the time-frequency characteristics of a recorded signal. This may affect the accuracy of PAM systems. To develop a PAM system capable of operating under numerous environmental conditions, one must account for the impact of propagation. An aural classifier developed at Defence R&D Canada (DRDC) has successfully been used for inter-species discrimination of cetaceans. The aural classifier achieves accurate results by using perceptual signal features that model the features employed by the human auditory system. The current work examines the relative impacts of signal-to-noise ratio (SNR) and propagation effects on the performance of the aural classifier. DRDC's pulse propagation model, Waveform Transmission Through a Channel (WATTCH), was used to simulate signals travelling through the ocean environment over ranges of 0–20 km. Noise was added to both these signals and the original signals, so that performance could be compared for three scenarios expected to decrease classifier performance: decreasing SNR, increasing propagation effects (frequency spreading, multipath, etc.), and combined SNR and propagation effects. In this presentation, the modeled results are compared to experimental data.

1:20

4pAO2. Bowhead whale localization and environmental inversion using asynchronous hydrophones. Graham A. Warner, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. Ste. A405, Victoria, BC V8P 5C2, Canada, gwarner@uvic.ca), David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada), and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper estimates bowhead whale locations and environmental properties using Bayesian inversion of the modal dispersion of whale calls recorded on asynchronous recorders in the shallow waters of the northeastern Chukchi Sea, Alaska. Bowhead calls were recorded on a Y-shaped cluster of seven autonomous ocean-bottom hydrophones, separated by up to 9.2 km. We use a warping time-frequency analysis to obtain frequency-dependent relative mode arrival times for nine frequency-modulated whale calls. Trans-dimensional inversion is applied to invert mode arrival times for the whale location, water sound-speed profile, subbottom layering and geoacoustic parameters, source instantaneous frequency (IF), relative recorder clock drifts, and residual error standard deviation, all with estimated uncertainties. Joint inversion of multiple calls is found to substantially reduce uncertainties on whale location, source IF, and clock drifts. Estimated whale location uncertainties are 30–160 m and clock drift uncertainties are 3–26 ms. The prior and posterior probability densities for environmental parameters are used to quantify transmission loss uncertainties corresponding to different levels of environmental knowledge, with applications to computing marine-mammal sound exposure levels.

1:35

4pAO3. Estimating source levels and depth distributions of calling bowhead whales using geoacoustic inversion. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Susanna Blackwell (Greeneridge Sci. Inc., Santa Barbara, CA), Katherine Kim (Greeneridge Sci. Inc., San Diego, CA), and A. Michael Macrander (Shell Exploration and Production Co., Anchorage, AK)

Every year during open-water season in 2008–2014, up to 40 passive acoustic recorders (DASARs) were deployed in the Beaufort Sea. Over the eight-year period, more than 3.1 million bowhead calls were localized. A vertical array was also deployed near DASARs in some years, enabling the matched-field geoacoustic inversion of bowhead whale calls to estimate the local acoustic propagation

environment. In turn, these environments were used to estimate the source level and calling depth distributions of nearly 50,000 calls between 2008 and 2014. The resulting source level distributions exhibited a stable mode near 160 dB re 1 uPa @ 1 m (rms), which proved robust to the year investigated, the whale call analysis method (manual or automated), and the type of acoustic propagation model (a simple empirically-derived "power-law" transmission loss model or a detailed waveguide propagation model). Depth distributions of calling whales were more sensitive to the environmental model used but always showed a mode at ~25 m depth. The evolution of the frequency distribution of calls over the years will also be discussed. This research highlights the importance of incorporating geoacoustic inversion and propagation modeling into marine mammal behavioral or density estimation studies. [Sponsored by Shell Exploration and Production Company.]

Contributed Papers

1:50

4pAO4. Applying double-difference methods for fine-scale acoustic tracking of controlled sources and sperm whales using a small aperture vertical array. Ludovic Tenorio-Hallé, Aaron M. Thode, Jit Sarkar (Scripps Inst. of Oceanogr., 1044 Loring St., San Diego, CA 92109, ludovictenorio@gmail.com), Chris Verlinden, Jeffrey D. Tippmann, William S. Hodgkiss, and William A. Kuperman (Scripps Inst. of Oceanogr., La Jolla, CA)

Ray propagation modeling can estimate a source's depth and range in a waveguide by exploiting multipath arrival information on a vertical array. However, environmental mismatch of the model, array tilt, and limited angular resolution of an array can yield highly scattered dive trajectories when ray tracing individual events. "Double-difference" methods have been used to localize earthquakes (Waldhauser and Ellsworth, 2000) and fin whales (Wilcock, 2012) by determining the location of multiple events relative to each other, rather than their absolute position. These same concepts can be reformulated into a "triple-difference" approach to track successive acoustic events on a single multi-hydrophone array. This method examines relative changes in the multipath arrival times and elevation angles over the course of a dive in order to establish a more robust track in terms of relative positions along the trajectory. Presented here are results of applying this new technique on both a towed source and sperm whales, using acoustic data recorded on a short aperture vertical array off the coast of Southern California in 4 km deep water. [Work supported by Office of Naval Research—Marine Mammals and Biology and Ocean Acoustics Program.]

2:05

4pAO5. Humpback whale-generated ambient noise levels provide insight into singers' spatial densities. Kerri D. Seger, Aaron M. Thode (Scripps Inst. of Oceanogr., 9331 Discovery Way, Apt. C, La Jolla, CA 92037, kseger@ucsd.edu), Jorge Urbán R., Pamela Martínez-Loustalot, M. E. Jiménez-López, and Diana López-Arzate (Universidad Autonoma de Baja California Sur, La Paz, Mexico)

Previous research has investigated whether diffuse ambient noise levels can be used to estimate relative baleen whale abundance in environments where their vocal activity dominates ambient noise levels. Presented here is an analytical model of ambient noise levels as generated by randomly distributed singing humpback whales. The model exploits earlier ones that derive ambient noise levels by assuming randomly distributed wind-driven breaking surface waves. Using a parametrized acoustic propagation environment and various assumptions about humpback whale singing behavior, the current model predicts ambient noise levels as a function of frequency and population size. It also predicts that the "sensitivity" of ambient noise levels to changes in population size should be relatively independent of an individual's singing behavior, but does depend strongly on the population's spatial density. Using visual survey and bottom-mounted recorder data from the 2013 and 2014 humpback whale breeding seasons off Los Cabos, Mexico, a generalized linear model estimated the sensitivity of song-generated ambient noise levels across several frequency bands, while compensating for diel cycles. The results indicate that whales generally space themselves evenly over a constantly expanding region, but may tolerate a slightly higher density as the number of participating singers increases.

2:20

4pAO6. Underwater acoustic vector sensor recording apparatus for soundscape measurements. Richard D. Lenhart and Jason D. Sagers (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, lenhart@arlut.utexas.edu)

An Autonomous Underwater Multi-Dimensional Acoustic Recorder (AUMDAR) for making directional soundscape measurements has been designed, constructed, and tested at The Applied Research Laboratories. The AUMDAR includes several three-dimensional transducers (acoustic vector sensor, gravitometer, accelerometer, and gyroscope), a multichannel recording unit (six channels, up to 96 kHz sampling rate and 24 bit resolution), and a battery power supply housed in a 90 m depth-rated pressure vessel. The apparatus can be deployed on the seafloor via an oceanographic tripod, suspended within the water column on a vertical mooring line, or tethered below a surface float. Laboratory-based acoustical performance characterization measurements including vector sensor on-axis receive sensitivity, directivity, and electrical noise floor are discussed. Field measurements recorded by the AUMDAR in Lake Travis, TX, and Haro Strait, WA, are also presented.

2:35–2:55 Break

2:55

4pAO7. Modelling the performance of fish tag monitoring stations on the Scotian Shelf. Danielle Moore and David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca)

The Ocean Tracking Network operates and maintains a continental shelf scale array of 256 bottom mounted fish-tag monitoring stations, spanning from the entrance of Halifax harbour to the Scotian Shelf break. These stations detect tagged keystone, commercially important, and endangered species as they migrate across this acoustic curtain, known as the Halifax line. The detection performance of each station is dependent on the local bathymetry, oceanography (sound speed profile variability), ambient noise level, and source depth distribution. At each station, local sound speed profiles from archived glider data were collected and sorted into representative groups. An Nx2D ray-trace model was used to calculate the transmission loss and relative strength of the noise field at fish tag frequencies (69 kHz) for each of the representative sound speed profiles at a number of stations across the array. The performance variability at each station and between stations is presented and compared to real detection data.

3:10

4pAO8. The effects of bottom sediments on the measured spectrum of seismic airgun pulses as a function of range. Bruce Martin, Jeff MacDonnell, and Loren Bailey (JASCO Appl. Sci., 32 Troop Ave., Ste. 202, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com)

Sound from seismic airgun pulses is well known to be a measurable component of the marine soundscape even hundreds of kilometers from the source vessel. The peak frequency of the pulses is normally below 100 Hz, and hence the concern is often raised that the airguns may mask the communications of large baleen whales. Recently, there have been reports of significant energy from seismic airguns in the range of 1–10 kHz which could mask calls from a wider range of marine life. Measurements and modeling of a variety of deep water environments are compared to show that environments with very hard bottom types support the long-range propagation of frequencies above 500 Hz from seismic airgun arrays.

Invited Papers

3:25

4pAO9. Broadband active acoustics for synoptic studies of marine ecosystem. Andone C. Lavery and Timothy K. Stanton (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

Marine ecosystems host a wide variety of organisms spanning many trophic levels, ranging from the smallest planktonic organisms that comprise the base of the marine food web, to large apex predator fishes, marine mammals, such as whales, and sea birds. Active acoustical systems that span a wide range of frequencies are required to quantitatively characterize these organisms, their distribution and abundance. In this presentation, we discuss the development and use of broadband acoustic systems that are optimized for studies of fish and zooplankton through use of spectral classification methods. For swimbladder-bearing fish, a low-frequency (1–6 kHz) broadband system is described that classifies fish according to the resonance of its swimbladder. For zooplankton, a high frequency (25–600 kHz) broadband system is described that classifies zooplankton either according the resonance of their gas (gas-bearing zooplankton only) or according to the transition between the Rayleigh and geometric scattering region (non-gaseous zooplankton). Because of the resolving power of the broadband signals, the echoes also tend to be non-Rayleigh, which has additional classification information. Applications of the spectral and statistical broadband acoustics methods to ecosystem research are given. [Work supported by the Office of Naval Research and the Woods Hole Oceanographic Institution.]

3:40

4pAO10. Fully utilizing the acoustic record for biological monitoring and ecological applications. John K. Horne, Ross Hytnen (School of Aquatic and Fishery Sci., Univ. of Washington, 1122 NE Boat St., Seattle, WA 98105, jhorne@uw.edu), Shari Maxwell (Pacific Northwest National Lab., Sequim, WA), Kenneth Ham (Pacific Northwest National Lab., Batelle, WA), Adam Maxwell (Pacific Northwest National Lab., Sequim, WA), and Jeff Condiotty (Simrad Fisheries, Lynnwood, WA)

Increasing demands to monitor marine ecosystems amplifies the need for efficient and effective characterization of organism abundances and distributions. Monitoring, processing, and archiving acoustic data may also be required in near-real-time when documenting environmental change or reporting impacts when meeting regulatory requirements. The Nekton Interactive Monitoring System (NIMS) is designed to address four challenges of ocean observation and monitoring: animal tracking, distribution characterization, regulatory thresholds, and data volume reduction. A Kalman filter is used to identify and link candidate targets into tracks. Backscatter measures are characterized using a suite of metrics (Sv, inertia, dispersion, Aggregation Index, evenness, % occupied) to quantify vertical distributions of pelagic organisms. If tracks cross pre-set exclusion ranges or regulatory thresholds of metrics are exceeded, then notifications are sent to trigger operational modification or mitigation. Kongsberg M3 sonar raw data acquisition rates of 11 Mb per second are reduced to 11 Mb per hour data storage, a 4 order of magnitude data savings. NIMS middleware can be deployed autonomously with instrument packages, remotely to telemeter data, network connected for real time monitoring, or used to process archived data. Example applications include ocean observatories and marine renewable energy environmental monitoring.

3:55

4pAO11. Application of acoustic technologies to study the temporal and spatial distributions of the Pacific hake (*Merluccius productus*) in the California Current System. Dezhang Chu, Rebecca Thomas (NOAA Fisheries, NWFSC, 2725 Montlake Blvd. E., Seattle, WA 98112, dezhang.chu@noaa.gov), Julia Clemons (NOAA Fisheries, NWFSC, Newport, Oregon), Sandy Parker-Stetter, John Pohl (NOAA Fisheries, NWFSC, Seattle, WA), Julia Clemons (NOAA Fisheries, NWFSC, Newport, Oregon), and Stephane Gauthier (Fisheries and Oceans Canada, Inst. of Ocean Sci., Sidney, BC, Canada)

Advances in acoustics technologies offer a remote and non-invasive sensing means to conduct fisheries acoustic surveys. Over the past two decades, joint US and Canada acoustic and trawl surveys on Pacific hake (*Merluccius productus*), one of the most important commercial fisheries off the West Coasts of the United States and Canada, have been conducted at the intervals of one to three years within the California Current System (CCS). In this presentation, the temporal and spatial distributions of Pacific hake resulting from these surveys spanning a period of nearly two decades will be presented. Challenges in converting the measured acoustic quantities to biological quantities, such as abundance and biomass, will be addressed, including uncertainties associated with mixed species, environmental parameters, and properties in fish morphology and anatomy. Issues related to transitions from single-species to ecosystem-based acoustic surveys will also be discussed.

Contributed Papers

4:10

4pAO12. High-frequency broadband acoustic backscatter from phytoplankton. Dylan L. DeGrâce (Oceanogr., Dalhousie Univ., 6299 South St., Halifax, NS B3H4R2, Canada, dylan.degrace@dal.ca) and Tetjana Ross (Inst. of Ocean Sci., Sidney, BC, Canada)

Current methods in phytoplankton detection and monitoring are often limited by low temporal and spatial resolution. In principle, the use of a high-frequency broadband acoustic system would be advantageous when used in conjunction with current methods; providing improvements both temporally and spatially. With this motivation, a high-frequency broadband active acoustic system has been developed and used in four separate trials to measure the backscatter from four morphologically-distinct species of phytoplankton. The morphologies studied include (1) a siliceous shelled cylinder, (2) a chain-forming siliceous shell cylinder, (3) a fluid-like spheroid, and (4) a soft-shelled spheroid; and whose sizes range from 10 to 60 μm. Organism cultures were insonified at frequencies between 0.75 MHz and 6.9 MHz giving a *ka* study range of 0.03–1.73. Volume scattering strength as it varies with *ka* is presented for each species and compared to potential scattering modelsdrawn from the zooplankton scattering literature. Modifications to the models or model parameters are discussed. Additionally, volume scattering strengths at multiple phytoplankton concentrations are

4p THU. PM

presented and compared to both chlorophyll-*a* estimates obtained from fluorometers and densities found via flow cytometry. The potential for a phytoplankton species-detection and monitoring system is discussed and evaluated.

4:25

4pAO13. Measurements of the acoustic properties of the seagrass *Posidonia oceanica*. Jay R. Johnson, Gabriel R. Venegas, Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), and Jean-Pierre Hermand (Université libre de Bruxelles, LISA - Environ. HydroAcoust. Lab, Brussels, Brussels Capital, Belgium)

A one-dimensional resonator technique was used to test the acoustic response of fresh leaves of *Posidonia oceanica* collected from both Crete, Greece and Sicily, Italy. The sound speed was inferred from resonance frequencies measured between 1 and 8 kHz. This sound speed is also compared with ultrasonic time-of-flight measurements between 1 and 4 MHz. Measurements of intact leaves as well as a homogenous tissue "soup" were made to investigate tissue properties and the role of leaf structure in sound propagation. This work expands on similar measurements, published previously by the authors, made on three other seagrass species *Thalassia testudinum, Halodule wrightii,* and *Syringodium filiforme*.

4:40

4pAO14. Bioacoustic absorption spectroscopy: An acoustical oceanography method for the study of marine ecosystems. Orest Diachok (Johns Hopkins Univ. APL, 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com)

Recent experiments have demonstrated the potential of the Bioacoustic Absorption Spectroscopy (BAS) method for measuring number densities vs. fish length and species when combined with concurrent trawling data, and studying vertical migrations and schooling behavior of fish at mesoscale dimensions.. Measurements of transmission loss between moving or stationary broadband source(s) and vertical array(s) of hydrophones separated by several (up to ~10) km permits inference of absorption coefficients versus frequency and depth, and association of absorption lines with the resonance frequencies of fish swim bladders. BAS is environmentally friendly, as it can be implemented with source levels $<$ 170 dB, is sensitive to fish throughout the water column, including near boundaries, and is not affected by *avoidance*. The most significant results of BAS experiments will be reviewed, and possible approaches for use of the BAS method for routine scientific study and fisheries surveys of marine ecosystems will be considered.

4:55

4pAO15. Active acoustic monitoring in extreme turbulence around marine renewable energy devices. Shaun Fraser (School of Eng., The Univ. of Aberdeen, Fraser Noble Bldg., Aberdeen AB24 3UE, United Kingdom, s.fraser@abdn.ac.uk), Benjamin Williamson, Beth E. Scott (School of Biological Sci., The Univ. of Aberdeen, Aberdeen, United Kingdom), and Vladimir Nikora (School of Eng., The Univ. of Aberdeen, Aberdeen, United Kingdom)

The advance of tidal energy technologies has created new demands for active acoustic monitoring in highly dynamic marine environments. An innovative data collection approach using the FLOWBEC multi-instrument platform has been developed to acoustically observe turbulence and ecological interactions in the challenging environments around turbine installations in the UK. Standard processing approaches for echosounder data are unsuitable in these sites because of the extreme variability in acoustic conditions due to strong tidal flows and complex wind-wave interactions. Novel techniques for identifying ecological targets (fish, diving seabirds, and marine mammals) and characterising the physical conditions have been developed which are functional even during extreme turbulence. Reliable target identification is achieved using scale-sensitive filtering, morphological characterization, and multifrequency analysis of EK60 echosounder data. Combining results with synchronized multibeam data and other observations gives new oceanographic and ecological insights into these environments. This study contributes novel methodological and processing concepts for acoustic analysis in challenging sites of emerging industrial importance. The results provide vital observations on the behaviour of marine species with clear applications for the analysis of environmental impacts of marine renewable energy technologies.

5:10

4pAO16. An evaluation of the frequency response of hydrocarbon droplets. Scott Loranger and Thomas C. Weber (Univ. of New Hampshire, Jere A Chase Ocean Eng., 24 Colovos Rd., Durham, NH 03824, sloranger@ccom.unh.edu)

Development of instrumentation to detect and quantify submerged oil droplets would provide researchers and oil spill responders with crucial information about the fate and movement of oil in the environment. By detecting oil droplets in the watercolumn it should be possible to trace surface sheens to their source and to determine the location and extent of plumes of oil at depth. Methods of detecting oil currently exist, for example, mass spectrometers and fluorometers; however, they are limited to detecting oil that is submeter range from the instrument. Using broadband high frequency (30–300 kHz) acoustic echosounders, it is possible to not only detect oil droplets from a greater distance (10s of meters for individual droplets, depending on the background noise) but to quantify the physical properties of the oil, including the size of droplets. Droplet size is an important factor in determining the likely location of submerged plumes and surface sheens, the rate of biodegradation and rise rate of oil. Laboratory measurements of the broadband response along with the sound speed, density and droplet size of crude oil, diesel, gasoline, and kerosene have been made. The frequency response of the droplets have been compared to models for the target strength of fluid filled spheres to verify the models, and to empirically derive adjustments if necessary. The data are also used to empirically estimate a detection range limit for different densities of droplets determined.

5:25

4pAO17. Time-difference-of-arrival localization of bowhead whales using asynchronous recorders. Graham A. Warner, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. Ste. A405, Victoria, BC V8P 5C2, Canada, gwarner@uvic.ca), and David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada)

This paper estimates bowhead whale locations and uncertainties using linearized Bayesian inversion of the time-difference-of-arrival (TDOA) of whale calls recorded on omni-directional asynchronous recorders in the Chukchi Sea, Alaska. A Y-shaped cluster of seven autonomous ocean-bottom hydrophones, separated by 0.5–9.2 km, was deployed for several months over which time their clocks drifted out of synchronization. Hundreds of recorded whale calls are manually annotated with time-frequency bounds and associated between recorders. The TDOA between all hydrophone pairs are calculated from filtered waveform cross-correlations and depend on the whale locations, hydrophone locations, relative recorder clock drifts, and an effective waveguide sound speed. The inversion estimates all of these parameters and their uncertainties as well as data error statistics using prior information to constrain the otherwise underdetermined problem. Whale location uncertainties are estimated to be approximately 100 m which allows tracking whales that vocalize repeatedly over several minutes. Estimates of clock drift rates are obtained from inversions of TDOA data over several weeks. The inversion is computationally efficient and suitable for application to large datasets of manually- or automatically detected whale calls.

Session 4pBA

Biomedical Acoustics: Imaging

Kausik Sarkar, Chair
George Washington University, 801 22nd Street NW, Washington, DC 20052

Contributed Papers

1:30

4pBA1. Simulation and Implication of a two-dimensional phased array flexible ultrasound system for tissue characterization. Zhihua Gan (Biomedical Eng., Stony Brook Univ., Rm. 212A, BioEng. Bldg., Stony Brook, NY 11794, zhihua.gan@stonybrook.edu), Brian Guo (Dynamic Res. Instruments LLC, Stony Brook, New York), Jiqi Cheng, and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., Stony Brook, NY)

It has been demonstrated that phased-array ultrasound can be used for configuring spatial focal zone and perform noninvasive characterization of trabecular bone quality. Such a sensor configuration can also be integrated into existing ultrasound triggering, data analysis, mux, and imaging system in a flexible ultrasound platform. The primary goal of this study is focused on the unique dual 2-D array transducer kit design with 1MHz center frequency and simulation to a flexible ultrasound system for trabecular bone assessment. Transmission (Tx) and receiving (Rx) transducers are designed as two identical 2-D array sensors, with elements arrayed in 27*27 as a square. Element width is 1.05 mm and kerf is 0.2 mm. Tx elements are divided into sub-blocks to excite ultrasound signal in sequence to decrease the system complexity while maintaining beam pattern properties by the signal processing procedure at Rx side. Appropriate delay modules are inserted into the process of exciting signal. The working procedure is simulated by FIELD II program. The gray-scale simulation of acoustic resolution and side lobes are analyzed. The theoretical lateral resolution (LR) is 2.436 mm, with 2.8 mm in simulation. Hann window based apodization process shows the reduced side lobes. The results suggested that the reduced array size (27*27) simplifies the design and manufacturing, but maintains the image/signal resolution.

1:45

4pBA2. A stable numerical method for wide-angle parabolic models of focused ultrasound. Joshua Soneson (Div. of Appl. Mech., US Food and Drug Administration, 10903 New Hampshire Ave., Silver Spring, MD 20993, joshua.soneson@fda.hhs.gov)

Wide-angle parabolic equations offer an attractive balance of utility, speed, and ease of implementation in the computation of sound beams in the frequency domain. These models are characterized by representing the pseudodifferential operator of the one-way Helmholtz equation with a Padé approximation, which is a quotient of polynomial operators, to which Crank-Nicolson-type numerical methods are easily applied. However, standard methods for determining the coefficients of these polynomials yield numerical schemes whose solutions exhibit oscillatory artifacts when source discontinuities are present, as is often the case for ultrasound transducers. In this work, two of three polynomial coefficients in a Padé approximation are used to obtain second order accuracy while the third coefficient is used to optimize stability. This results in a numerical method which effectively damps the spurious oscillations but retains the wide-angle capability, bringing to medical ultrasound a simple way to rapidly compute shallowly focused or steered beams.

2:00

4pBA3. Development of a nonlinear model for the pressure dependent attenuation and sound speed in a bubbly liquid and its experimental validation. Amin Jafari Sojahrood (Dept. of Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B2K3, Canada, amin.jafarisojahrood@ryerson.ca), Qian Li, Mark Burgess (Mech. Eng., Boston Univ., Boston, MA), Raffi Karshafian (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada), Tyrone Porter (Mech. Enginnering, Boston Univ., Boston, MA), and Michael C. Kolios (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada)

Presence of the MBs in the sound field increases the attenuation of the medium and changes the sound speed. A detailed knowledge about the attenuation of the medium is critical for controlling and optimizing the behavior of the MBs in applications. However, existing models of ultrasound attenuation in bubbly mediums are based on linear approximations (low amplitude MB oscillations) and thus are not valid in many regimes used in applications. A model to calculate the nonlinear attenuation and sound speed is developed by deriving the complex and real part of the wave number from the Calfish model. The predictions of the model were validated by measuring the attenuation and sound speed of dilute monodisperse MB solutions (5000 microbubbles/ml) with median diameter of 5.2 and 9.8 μm using acoustic pressure range of 10–130 kPa. The attenuation of the medium was calculated by numerically solving the radial oscillations of the MB and incorporating it in the attenuation model. Predictions of the model were in good agreement with the experimental results. As the acoustic pressure increased, the attenuation and maximum sound speed of the medium increased from 5 dB/cm to 12 dB/cm and 1500 to ~1530 m/s, respectively.

2:15

4pBA4. A stethoscope for the knee: Investigating joint acoustical emissions as novel biomarkers for wearable joint health assessment. Omer T. Inan, Sinan Hersek, Caitlin N. Teague, Hakan Toreyin, Hyeon K. Jeong (Elec. and Comput. Eng., Georgia Inst. of Technol., Technol. Square Res. Bldg., 85 Fifth St NW, Ste. 412, Atlanta, GA 30308, inan@gatech.edu), Michael L. Jones, Melinda L. Millard-Stafford, Geza F. Kogler, and Michael N. Sawka (Appl. Physiol., Georgia Inst. of Technol., Atlanta, GA)

Each year, millions of Americans endure knee injuries, ranging from simple sprains to ligament tears requiring surgical intervention. Our team is investigating wearable rehabilitation assessment technologies for patients recovering from knee injuries based on the measurement and analysis of the acoustical emissions from the knees. Using miniature electret microphones combined with piezoelectric sensors placed on the surface of the skin at the knee, we measure the sounds from the joint as subjects perform basic flexion/extension exercises and standardized sit-to-stand protocols. We then analyze the consistency of the knee acoustical emissions in the context of the activity, and the angle of the joint, to quantify the health of the joint. We have found, in early pilot studies, promising results differentiating the healthy versus injured knee, and longitudinal changes progressing from acute injury and recovery following rehabilitation. We have also determined

that, in healthy subjects, the pattern of acoustic emissions is consistent within several repetitions of a movement, and for multiple recordings throughout the day ($r > 0.88$). Knee acoustic emissions combined with angle measurements provide promising in-depth information regarding joint health and exciting new opportunities for personalized rehabilitation protocols following injury.

2:30

4pBA5. Photoacoustic imaging of muscle oxygenation during exercise. Clayton A. Baker, Nashaat Rasheed, Parag V. Chitnis, and Siddhartha Sikdar (BioEng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, cbaker6@masonlive.gmu.edu)

Monitoring muscle hemodynamics and oxygenation is important for studying muscle function and fatigue. Current state-of-the-art for noninvasive oximetry is near-infrared spectroscopy (NIRS), which provides relative oxygen saturation (SO_2) with good sensitivity but has poor spatial resolution and sensing depth, and does not provide anatomical context. In this study, we demonstrated the utility of a dual-modality imaging system that could generate co-registered ultrasound (US) and photoacoustic (PA) images for real-time, functional imaging of human muscle. The system consisted of a wavelength-tunable pulsed laser (Opotek) integrated with a research ultrasound system (Verasonics). US images provided anatomical context and the PA images acquired at 690 nm and 830 nm were processed to estimate SO_2 during a sustained isometric contraction and return to rest using a protocol approved by our Institutional Review Board. PA-based SO_2 was compared to measurements acquired from a commercially available NIRS oximeter under the same conditions. Preliminary results showed good qualitative agreement between SO_2 dynamics observed via PA-based approach and NIRS, with contraction coinciding with an exponential decay in SO_2, and relaxation with a return to original levels. This *in vivo* study demonstrated the feasibility of PA imaging for measuring temporally and spatially resolved muscle oxygenation during functional tasks.

2:45

4pBA6. Clinical results of ultrasound bladder vibrometry for assessment of bladder compliance. Mahdi Bayat, Mathew Cheong, Max Denis, Viksit Kumar (Physiol. and Biomedical Eng., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu), Mohammad Mehrmohammadi (Biomedical Eng., Wayne State Univ., Detroit, MI), Adriana Gregory, Ivan Nenadic (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN), Douglas Husmann, Lance Mynderse (Urology, Mayo Clinic, Rochester, MN), Azra Alizad, and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Healthy human bladder is a viscoelastic shell capable of expanding to allow storage of urine at low pressures. The ability to maintain low pressures as the urine volume increases is called bladder compliance. Compliance is a key in proper functioning of urinary system. Spinal cord injuries and other neurogenic diseases can cause deterioration in bladder expansion capability (e.g., through excessive growth of fibrotic tissue), which in turn can increase the rigidity of the wall and reduce bladder compliance. Urodynamic study (UDS) is the clinical standard for evaluation of the bladder compliance through observation of bladder pressure-volume behavior. However, this method is costly and invasive. Ultrasound bladder vibrometry (UBV) has been introduced as noninvasive technique for evaluation of bladder viscoelasticity through excitation of the bladder using ultrasound radiation force and tracking the resulting Lamb waves. Here, we present the results of UBV from 44 patients undergoing concurrent UDS and UBV studies. Our results show that elasticity parameters obtained from the UBV closely correlate with the UDS pressure-volume data with a median Pearson's correlation value of more than 0.8. These results prove the potential utility of UBV as an alternative noninvasive method for bladder compliance assessment. [Work supported by NIH grant DK99231.]

3:00

4pBA7. Phantom study on the detectability of micro-tumors in breast tissue using high-frequency ultrasound. Nicole Cowan (BioTechnol., Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, ncowan18@gmail.com), Zachary A. Coffman (Biology, Utah Valley Univ., Orem, UT), Robyn K. Omer (Botany, Utah Valley Univ., Orem, UT), Benjamin F. Finch (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The ability to detect malignant tissue in surgical margins during breast cancer surgery would reduce the risk of local recurrence and the need for subsequent surgeries to remove residual cancer. A surgical study conducted by Utah Valley University with the Huntsman Cancer Institute showed that high-frequency ultrasound (20–80 MHz), and the parameters peak density (number of spectral peaks and valleys from 20 to 80 MHz) and attenuation, are sensitive to breast tissue pathology. Pathology results from this surgical study showed that many margin specimens contained micro-tumors measuring 1 mm in diameter or smaller. The present study's objective was to determine the sensitivity of high-frequency ultrasound to these micro-tumors using phantoms. Phantoms were created from distilled water, agarose powder, 10X TBE stock solution, and 390–925μm diameter polyethylene microspheres to simulate breast tumors. Microspheres were embedded in phantoms singularly and in clusters of 3–12 microspheres. Pitch-catch measurements were acquired using large (6.35 mm diameter) and small (1.5 mm diameter) 50-MHz transducers, a high-frequency ultrasound system, and glycerol as the coupling agent. Both large and small transducers were sensitive to single microspheres and microsphere clusters across all microsphere diameters. The phantom results confirm the sensitivity of high-frequency ultrasound to breast cancer micro-tumors and validate the surgical study.

3:15

4pBA8. High-frequency ultrasonic measurements of ischemia and revascularization in mice. Michaelle A. Cadet (Biology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, michaelle.alexandra@gmail.com), Andrea N. Quiroz (Nursing, Univ. of Miami, Miami, FL), Andrew Chappell (Dentistry, Univ. of Louisville, Louisville, KY), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The ability to rapidly determine the degree of vascularization in small animals *in vivo* would provide a useful characterization tool for regenerative medicine. This study's objective was to determine if ultrasonic material property measurements in the 10–100 MHz range could be used as a vascularization assay for small animals. The study was performed at the Ludwig Boltzmann Institute for Experimental and Clinical Traumatology (Vienna, Austria), where the femoral artery in one hind limb of each of sixteen mice was ligated and tested over a time period of eight days. Eight of the ligated limbs were treated with vascular endothelial growth factor (VEGF). The remaining eight ligated limbs were allowed to grow ischemic. The unligated limbs were controls. Wave speed, attenuation, waveform coda amplitude, and spectral measurements were acquired using a high-frequency ultrasound system. Ischemic limbs displayed a steady decrease in wave velocity over the test period as compared to VEGF-treated limbs. Coda amplitude increased for ischemic limbs, but decreased and then returned to normal in VEGF-treated limbs. No trends were observed for either attenuation or spectral peak density. The results indicate that high-frequency ultrasonic measurements may provide an added dimension to small animal imaging methods for detecting revascularization.

3:30–3:45 Break

3:45

4pBA9. Effect of mammographic breast density on high-frequency ultrasonic parameters used to evaluate surgical margins. Zachary A. Coffman (Biology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, zachary.a.coffman@gmail.com), Nicole Cowan (BioTechnol., Utah Valley Univ., Orem, UT), Robyn K. Omer (Botany, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Breast density, the proportion of connective tissue versus fat tissue in the breast, is typically determined using mammography. Women with higher breast density are four to five times more likely to develop breast cancer than women with lower densities. A surgical study performed by Utah Valley University with the Huntsman Cancer Institute showed that high-frequency ultrasound (20–80 MHz), and the parameters peak density (number of spectral peaks and valleys from 20 to 80 MHz) and attenuation, are sensitive to breast tissue pathology. This study also showed that breast density had no effect on peak density across the entire breast density range while attenuation increased 2X from entirely fat to extremely dense. The present study's objective was to determine the effect of breast density on these parameters using histology mimicking phantoms. Phantoms were created from distilled water, agarose powder, 10X TBE stock solution, and polyethylene microspheres and fibers to simulate breast tissue histology. Phantoms were produced with microspheres only, fibers only, and a combination of microspheres and fibers. Pitch-catch measurements were acquired using a high-frequency ultrasound system and glycerol as the coupling agent. The phantom results validate the surgical study, showing attenuation increasing with inclusion concentration but no significant change in peak density.

4:00

4pBA10. Interpreting attenuation at different excitation amplitudes to estimate strain-dependent interfacial rheological properties of monodisperse contrast microbubbles. Kausik Sarkar, Lang Xia (George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu), and Tyrone Porter (Boston Univ., Boston, MA)

Broadband attenuation of ultrasound measured at different excitation pressures being different raises a serious theoretical concern, because the underlying assumption of linear and independent propagation of different frequency components nominally requires attenuation to be independent of excitation. This issue is investigated by examining ultrasound attenuation through a monodisperse lipid coated microbubble suspension measured at four different acoustic excitation amplitudes. We use the attenuation data to determine interfacial rheological properties (surface tension, surface dilatational elasticity and surface dilatational viscosity) of the encapsulation according to three different models. Although different models result in similar rheological properties, attenuation measured at different excitation levels (4–110 kPa) leads to different values for them; the dilatation elasticity (0.56 N/m to 0.18 N/m) and viscosity (2.4×10^{-8} Ns/m to 1.52×10^{-8} Ns/m) both decrease with increasing pressure. Numerically simulating the scattered response, nonlinear energy transfer between frequencies are shown to be negligible, thereby demonstrating the linearity in propagation and validating the attenuation analysis. There is a second concern to the characterization arising from shell properties being dependent on excitation amplitude which is not a proper constitutive variable. It is resolved by arriving at a strain-dependent rheology for the encapsulation.

4:15

4pBA11. Quantitative microultrasound characterization of gastrointestinal tissue for ultrasound capsule endoscopy. Holly Lay (School of Eng., Univ. of Glasgow, Glasgow G12 8QQ, United Kingdom), Benjamin F. Cox (Ninewells Hospital and Med. School, Dundee, United Kingdom), Christine E. Demore (School of Eng., Univ. of Glasgow, Glasgow, United Kingdom), Gabriel C. Spalding (Illinois Wesleyan Univ., Bloomington, IL), and Sandy Cochran (School of Eng., Univ. of Glasgow, Glasgow, United Kingdom, sandy.cochran@glasgow.ac.uk)

Gastrointestinal (GI) disease progression is often characterized by cellular and architectural changes within the mucosal and sub-mucosal layers. One relevant disorder, Barrett's esophagus, is of particular interest as a recognized predictor of esophageal cancer. To enhance the clinical ability to detect cellular changes deeper in tissue and earlier, we are exploring quantitative microultrasound) techniques in healthy *ex vivo* porcine GI tissue for implementation in ultrasound capsule endoscopy. A single-element, piezocomposite mUS transducer operating at fc = 47.7 MHz was used to obtain pulse-echo images of *ex vivo* porcine gastroesophageal samples, which were bisected and mechanically scanned along the long GI axis. Selected samples were mechanically separated to isolate the upper mucosal layers from the underlying muscle layers and placed on a known agar substrate to calculate the attenuation coefficient of the tissue. All other sample data were digitally segmented. Reflectivity data from the top 50, 100, and 250 μm of tissue were analysed to assess the effect of variable surface density on the calculated acoustic impedance. The entire thickness of the segmented tissue was then used to calculate position-dependent backscatter coefficient (BSC) along with intra- and inter-sample variability for use as a baseline from which to make further quantitative advances.

4:30

4pBA12. Prediction of multivalued waveforms in media with power-law attenuation. John M. Cormack and Mark F. Hamilton (Appl. Res. Lab., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

The lossless Burgers equation predicts a multivalued waveform beyond a certain propagation distance. Inclusion of thermoviscous attenuation, which increases as frequency squared, prevents the occurrence of multivalued waveforms. The same is true for any attenuation law that is proportional to frequency raised to an exponent greater than unity. For exponents less than unity the situation is less clear. For example, when attenuation is constant with frequency (exponent equal zero) there is a critical value of the attenuation coefficient below which a multivalued waveform is predicted and above which it is not. To investigate the prediction of multivalued waveforms for power-law attenuation with exponents between zero and unity, a Burgers equation with the loss term expressed as a fractional derivative is used [Prieur and Holm, J. Acoust. Soc. Am. **130**, 1125 (2011)]. Transformation of the equation into intrinsic coordinates following Hammerton and Crighton [J. Fluid Mech. **252**, 585 (1993)] permits numerical solutions to be obtained that are used to determine the parameter space in which initially sinusoidal plane waves are predicted to evolve into multivalued waveforms for power-law attenuation with exponents less than unity. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:45

4pBA13. Evaluating lymph node status with high-frequency ultrasound during breast conservation surgery. Amy A. LaFond (Biology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, fairbrother.aa@gmail.com), Caitlin Carter (BioTechnol., Utah Valley Univ., Orem, UT), Dolly A. Sanjinez (Biology, Utah Valley Univ., Orem, UT), Robyn K. Omer (Botany, Utah Valley Univ., Orem, UT), Leigh A. Neumayer (Surgery, Univ. of Arizona, Tucson, AZ), Rachel E. Factor (Pathol., Univ. of Utah, Salt Lake City, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency (20–80 MHz) ultrasound was used to evaluate 78 lymph nodes from 39 patients to identify metastatic breast cancer during breast conservation surgery. Point measurements were collected from resected lymph nodes in both through-transmission and pulse-echo modes using 50-MHz, 6.35-mm diameter, single-element transducers. Attenuation and peak density (the number of peaks and valleys in a specified frequency band) were calculated from the ultrasonic waveforms and power spectra, respectively. The two parameters were additionally combined to perform a multivariate analysis. Fisher's exact test was used to determine the accuracy, sensitivity, and specificity of each parameter and the multivariate analysis for detecting malignant lymph nodes. The multivariate analysis showed the greatest statistical measures, with an 83.3% accuracy, 87.5% sensitivity, 82.9% specificity, and a p-value of 0.000078 (high statistical significance). The results demonstrate that high-frequency ultrasound provides very good sensitivity and specificity for malignant lymph nodes in the breast, and that high-frequency ultrasound is a viable, prospective method to be used as a rapid, intraoperative, and potentially*in vivo*diagnostic tool by surgeons for a wide range of soft tissue cancers.

5:00

4pBA14. Detecting changes in the cytoskeletal structure of malignant pancreatic cells using high-frequency ultrasound. Caitlin Carter (Bio-Technol., Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, caitlin.carter03@gmail.com), Ashley Behan, Dolly A. Sanjinez, Amy A. LaFond (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Traditional classification of breast cancer subtype is determined based on protein expression and genetic profile. Determining the molecular subtype of breast cancer can make prognosis more accurate and help to personalize treatment. More aggressive subtypes of breast cancer, such as basal-like and Her2 +, have mutations that alter the protein regulation of the cell cytoskeleton. These cytoskeleton modulations can enable the cell to become more mobile and metastasize to other parts of the body. High-frequency ultrasound (10–100 MHz) has previously been studied for the real-time diagnosis of malignant tissue in breast conservation surgery. High-frequency ultrasound also has the potential of determining the molecular subtype of breast cancer. The objective of this work was to determine if chemically induced changes in the cytoskeleton of cancer cell lines are able to be detected using high-frequency ultrasound. Cell cultures of a human pancreatic carcinoma cell line (panc-1) were grown in monolayers and then treated with sphingosylphosphorylcholine (SPC), a bioactive lipid that rearranges the keratin components of the cytoskeleton. Pulse-echo measurements of the cultures were taken over a period of one hour. The results of this work demonstrate that high-frequency ultrasonic spectra are sensitive to the cytoskeleton changes induced by SPC.

THURSDAY AFTERNOON, 26 MAY 2016 — SOLITUDE, 1:30 P.M. TO 6:00 P.M.

Session 4pMU

Musical Acoustics: Session in Honor of William J. Strong

Thomas Rossing, Chair
Stanford University, 26464 Taaffe Rd., Los Altos Hills, CA 94022

Chair's Introduction—1:30

Invited Papers

1:35

4pMU1. The impact of William J. Strong on the acoustics program at Brigham Young University. Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

The early years of acoustics at Brigham Young University (BYU) include names such as Eyring, Fletcher, and Knudsen. However, for nearly 35 years, Bill Strong was the face of acoustics at BYU. As a faculty member, he contributed significantly to the maintenance and strengthening of acoustics at BYU, influencing many lives along the way. One of those lives influenced was the author. Beyond his positive influence in shaping many lives, Bill also contributed significantly to acoustics research throughout his career. Much of his work centered on the acoustics of speech and music. His text, *Music, Speech, Audio*, has been used for decades to introduce many students to the field of acoustics and has been the text for a popular course on the subject at BYU. This presentation will overview some of Bill's many contributions, both in acoustics and otherwise, with an emphasis on some of the work carried out by the author and Bill on the acoustics of the clarinet. That work includes a study of the interactions associated with the combined player-clarinet system, as well as an investigation of the directivity of the sound radiated from the clarinet.

2:00

4pMU2. How well can a human mimic the sound of a trumpet? Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, 156 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@utah.edu)

The length of an uncoiled trumpet horn is more than 2 m, whereas the length of a human supraglottal airway is about 17 cm. The vibrating lips and the vibrating vocal folds can produce similar pitch ranges, but the resonators have vastly different natural frequencies due to the more than 10:1 ratio in airway length. Humans rarely attempt to tune more than one resonance to a harmonic because of the disparity between harmonic spacing and airway resonances spacing, but the brass-like timbre of a human "call" or "belt" can mimic that of a trumpet. A one-dimensional Navier-Stokes solution of non-steady compressible flow in soft-walled airways is used to show the similarities and differences in time-domain wave propagation and resulting flow and pressure frequency spectra along the airways. One striking structural similarity between the human instrument and the brass instrument is the shape of the airway directly above the sound source, known as the mouthpiece for brass and the epilarynx tube for the human airway. It plays a major role in shaping the source spectrum.

2:25

4pMU3. William J. Strong's musical instrument models. James W. Beauchamp (School of Music and Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824, jwbeauch@illinois.edu)

After studying acoustics with Harvey Fletcher at Brigham Young University (BYU) in the 1950s, Bill joined the computer music sound analysis/synthesis group led by Melville Clark, Jr., at the Massachusetts Institute of Technology in 1959. He presented his first musical acoustics paper, on clarinet tone synthesis, at the fall, 1964 ASA meeting in Austin. This was followed by a paper on oboe synthesis at the spring, 1965 meeting in Washington. These papers were elaborated on in two wind instrument synthesis papers he and Clark published in JASA in 1967. The pioneering method was called *spectral envelope synthesis*, although temporal envelope functions were also incorporated in the model. Parameters for this complex synthesis method were obtained from time-variant spectrum analyses done by David Luce (Ph.D. thesis, MIT, 1963). Later back at BYU, in the late 1970s and 1980s, Bill and his students began analyzing acoustic structures of wind instruments, culminating with Michael Thompson (M.S. thesis, BYU, 2000) and Bill publishing results on measurement and simulation of nonlinear propagation in a trombone in JASA in 2001. This clearly demonstrated that nonlinear propagation of waveforms in a pipe can produce very significant effects on the "brassiness" of a brass instrument's output sound.

2:50

4pMU4. Acoustical factors affecting the playability of brass wind instruments with side holes. D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk)

Almost all modern instruments of the brass wind family radiate sound through a single terminating bell. From the Middle Ages until the second half of the nineteenth century, lip-excited wind instruments with side holes were also in frequent use in both secular and sacred musical ensembles. The acoustical properties of labrasones with side holes, such as the cornetto, the serpent, the keyed bugle, and the ophicleide, resemble in may ways those of conventional brass instruments such as the trombone, but also have features which recall woodwind instruments like the oboe and saxophone. This paper presents some recent studies on the playability of labrasones with side holes, acknowledging the outstanding contributions of William J. Strong and his collaborators to the experimental study and modelling of both woodwind and brass instruments.

3:15

4pMU5. Acoustics research at Brigham Young University during the late 1960s. George Plitnik (Phys., Frostburg State Univ., 120 Compton Hall, Frostburg, MD 21532, gplitnik@frostburg.edu)

As Dr. Strong's first Ph.D. student in musical acoustics (I actually arrived one semester before he did), he took on the task of reorganizing the musical acoustics program as well as adding a research division in speech. The fact that these programs were, and continue to be, successful is a tribute to Dr. Strong's dedication and hard work.

3:40–3:55 Break

3:55

4pMU6. Conical reed instruments: A hierarchy of parameters. Jean Kergomard, Philippe Guillemain, Fabrice Silva, and Christophe Vergez (CNRS-LMA, 31 Chemin Joseph Aiguier, Marseille 13402, France, kergomard@lma.cnrs-mrs.fr)

Some aspects of sound production by wind musical instruments are rather well understood. A minimum model for reed cylindrical instruments was proposed in 1983 by Mc Intyre *et al*. When the reed dynamics and resonator losses are ignored, the mouthpiece pressure is a square signal. Three primary parameters are necessary: (i) the mouth pressure; (ii) the "valve" parameter (based on the reed opening and its stiffness); (iii) the length of the cylinder. The model gives a simplified shape of the waveform; the playing frequency and the amplitude are rather well predicted. For further spectrum details, it is necessary to add losses, therefore a secondary parameter, the radius. Furthermore other parameters influence the spectrum: reed dynamics, toneholes, vocal tract.... What happens for conical reed instruments? Considering the waveform of the mouthpiece pressure, a fourth primary parameter is the length of the missing part of the truncated cone. Similarly to cylindrical instruments, a secondary parameter is related to the losses depending either on the input or the output one, or on the apex angle: one of these three parameters allows the determination of the two others. A discussion is proposed concerning the different kinds of conical instruments.

4:20

4pMU7. High-resolution directivities of musical instruments and the human voice: Recent research collaborations of Bill Strong. Timothy W. Leishman, K. J. Bodon, and Jennifer K. Whiting (Phys. and Astronomy, Brigham Young Univ., N247 ESC, Provo, UT 84602, twleishman@byu.edu)

Over the past few years, Bill Strong has worked with other researchers at Brigham Young University to develop a feasible method to assess high-resolution directivities of played musical instruments and the human voice. The approach has produced numerous directivity balloons with 5-degree angular resolution and variable bandwidths, meeting the requirements of the AES56-2008 standard for loudspeaker measurements. This presentation will give an overview of the method, its validation, and some interesting results for several musical instruments and talkers. Data from the measurements are intended to enhance theoretical and practical efforts of professionals seeking to better understand and work with these live sources of sound.

4:45

4pMU8. Acoustics in a physics program in the 1990s. Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 49504, dludwigs@kettering.edu)

To a recent graduate from a liberal arts college with majors in physics and music, Dr. Bill Strong presented a unique opportunity to work on a graduate degree in physics, with research in musical acoustics, using a computational approach. The historical tradition of acoustics within the discipline of physics was important to that young graduate student then, in the 1990s, and continues to be important to this member of academia twenty years later. The work of Bill Strong in shaping and sustaining acoustics at Brigham Young University was at a critical juncture. This presentation of the contemporary setting and my mentor's approach will offer a grad student's perspective, through an appreciative filter of years of experience as a professor.

5:10

4pMU9. From musical acoustics to noise control. David C. Copley (Caterpillar, PO Box 1875, Peoria, IL 61656, Copley_David_C@cat.com)

This presentation is a personal reflection of the influence Bill Strong had on the author's career in noise control. From first encounters with the decibel to experimental and numerical research in trombone acoustics, from simple textbook quizzes to demanding 20-page single-problem essays, the author shows the influence Bill Strong had on his education in acoustics which became the bedrock for a career in machinery noise control. The presentation will cite specific example of concepts, analyses and techniques found in musical acoustics—first encountered by the author in his studies with Bill Strong—and similarities to industrial noise control situations. The examples will demonstrate how an education in musical acoustic translates to practical applications in noise control engineering.

5:35

4pMU10. The effect of inharmonicity in piano tones. Brian E. Anderson and William J. Strong (Dept. of Phys. and Astronomy, Brigham Young Univ., BYU, N283 ESC, Provo, UT 84602, bea@byu.edu)

This presentation will review research conducted to determine the effect of inharmonic partials on the pitch of piano tones [JASA, **117**, 3268–3272 (2005)]. Synthetic piano tones were created based on recordings of an upright piano. One set of tones were made to have inharmonic partials that matched the partial frequencies of the piano tone recording, whereas another set of tones consisted of harmonic partials. Listeners compared a synthetic, inharmonic piano tone to a set of synthetic, harmonic piano tone with varying fundamental frequencies. They were asked to select the harmonic tone whose pitch best matched that of the inharmonic tone. The pitch of the inharmonic tones was perceived to be sharp relative to the harmonic tones (sharp in terms of the comparison of their respective fundamental frequencies).

Session 4pNS

Noise, ASA Committee on Standards, Animal Bioacoustics, Engineering Acoustics, and Physical Acoustics: Wind Turbine Noise II

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118

Kenneth Kaliski, Cochair
RSG Inc, 55 Railroad Row, White River Junction, VT 05001

Robert D. Hellweg, Cochair
Hellweg Acoustics, Wellesley, MA 02482

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Drive, Champaign, IL 61821

Contributed Papers

1:30

4pNS1. Monitoring the acoustic effects of pile driving for the first offshore wind farm in the United States. Arthur E. Newhall, Ying T. Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 210 Bigelow Lab, MS #11, Woods Hole, MA 02543, anewhall@whoi.edu), James F. Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Kathy Vigness-Raposa, Adam Frankel, Jennifer Giard (Marine Acoust., Inc., Middletown, RI), Dennis R. Gallien, Jamey Elliot (HDR, Inc., Athens, AL), and Tim Mason (Subacoustech Environ. Ltd., Bishops Waltham, Hampshire, United Kingdom)

The Block Island Wind Farm, the first offshore wind farm in the United States, consists of five turbines in water depths of approximately 30 m. The turbines have a jacket type substructure with piles driven to the bottom to pin the structure to the seabed. A number of acoustic sensors were deployed to monitor the acoustic properties of the pile driving activity. The acoustic sensor systems consisted of an eight element towed hydrophone array, two fixed moorings with four hydrophones each, and a fixed sensor package for measuring particle velocity. The towed array was towed from 1 to 8 km from the pile driving location. The fixed moorings were deployed at 7.5 and 15 km from the pile location. The particle velocity sensor package was deployed at 500 m from the pile driving location. This sensor package consisted of a three-axis geophone on the seabed and a tetrahedral array of four low sensitivity hydrophones at 1 m from the bottom. Data collected on these sensor systems will be presented. Acoustic levels and particle velocity observations will be compared and their implications in the context of effects on marine life will be discussed. [Work supported by Bureau of Ocean Energy Management (BOEM).]

1:45

4pNS2. Guidelines for developing regulations for acoustic impact, based on the stage of operation of wind farms in Chile. Elias N. Montoya and Ismael P. Gomez (Universidad Tecnológica de Chile INACAP, Brown Norte 290, Ñuñoa, Santiago 7750000, Chile, enmontoyag@gmail.com)

Wind farms manifest different sonic characteristics than other sources, since the noise emitted depends on the wind which turns the blades and varies constantly, being able to tonal components, amplitude modulated, and low frequency noise. For these reasons, the international community has developed specific regulations to assess the acoustic impact associated. Nevertheless, the present Chilean noise regulations are not adjusted properly to the situation described above. Considering the exponential increase experienced by this energy source in recent years in Chile, the development of regulations for such features in the stage operation is suggested so that the noise impact can be properly addressed and quantified. Guidelines for the development of an eventual specific Chilean regulation will be proposed by analyzing and comparing international standards. From this work, we will determine the maximum noise levels, methodologies and other suggestions for the proposed target issues.

2:00

4pNS3. Measurements of infrasound blade pass frequencies inside multiple homes using narrowband analysis. Andy Metelka (Sound and Vib. Solutions Canada, Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Previous measurements in homes near wind turbines indicate higher pressure levels below 10Hz than audible pressure levels measured at the same time and location (Dooley and Metelka, ASA **20**, 2013). Long term measurements of Infrasound pressures appear inside multiple homes as wind speed and wind direction vary. Measuring Narrowband signatures identify fingerprints of rotational components of wind turbines. The measurements directly correlate how blade pass frequencies appear and disappear everywhere as wind changes. Data from four homes were measured and broadband infrasound levels from wind are compared to tonal infrasound blade pass frequencies In both cases, broadband infrasound and blade-to-tower pressures increase with wind and advanced signal processing techniques were used tracking signatures buried in noise.

2:15–4:15 Panel Discussion

Session 4pPAa

Physical Acoustics: General Topics in Physical Acoustics I

Kevin M. Lee, Chair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Contributed Papers

1:00

4pPAa1. Origin of negative density. Sam Hyeon Lee (Phys., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea), samlee@yonsei.ac.kr) and Oliver B. Wright (Div. of Appl. Phys., Hokkaido Univ., Sapporo, Japan)

We present a new physical interpretation for the effective densities. We introduce the idea of hidden force: the effective density is negative when the hidden force is larger than, and operates in antiphase to the applied force. We demonstrate this picture for some established acoustic metamaterials with elements based on membranes, springs, and masses. The hidden force for membrane-based acoustic metamaterials, for instance, is the force from the membrane tension. We also explain the analogous concepts for pure mass-and-spring systems, in which case, the hidden force can arise from masses and springs fixed inside other masses. This new picture provides a powerful tool for conceptual understanding and design of new acoustic metamaterials, and avoids common pitfalls involved in determining the effective parameters of such materials.

1:15

4pPAa2. Origin of negative modulus. Sam Hyeon Lee (Phys., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, samlee@yonsei.ac.kr) and Oliver B. Wright (Div. of Appl. Phys., Hokkaido Univ., Sapporo, Japan)

We present a new view point for effective moduli of acoustic metamaterials. We introduce the concept of hidden source of volume: the effective modulus is negative when the volume of fluid injected from the hidden source is larger than, and operates in antiphase to, the volume change of the unit cell that would be obtained in its absence. We demonstrate this ansatz for some established acoustic metamaterials with elements based on Helmholtz resonators and side holes. The hidden source for a Helmholtz-resonator-based metamaterial is the extra air volume injected from the resonator cavity. The hidden sources—more aptly termed hidden expanders of displacement in this case—can arise from light rigid trusses coupled to extra degrees of freedom for mechanical motion such as the case of coupling to masses that move at right angles to the wave-propagation direction.

1:30

4pPAa3. A broadband sound absorber based on lossy acoustic rainbow trapping metamaterials. Tuo Liu, Jie Zhu, and Li Cheng (Mech. Eng., The Hong Kong Polytechnic Univ., Rm. FJ610, 11 Yuk Choi Rd., Hung Hom, Kowloon, Hong Kong, toneliutuo@gmail.com)

Acoustic rainbow trapping (ART) metamaterials offer spatial-spectral modulation and intensive trapping of broadband sound, with strong acoustic dispersion that is absent in naturally occurring materials. However, thermal and viscous losses stemming from the acoustic boundary layers within narrow resonance regions of ART metamaterials has not been thoroughly studied and effectively utilized. Here, we would like to propose a lossy model of ART metamaterials. By taking advantage of the thermal and viscous losses, such lossy metamaterials can effectively work as a broadband sound absorber. A rigid surface perforated with very narrow subwavelength holes that can provide boundary-layer losses and have gradient depth along the propagation direction is constructed to mimic the lossy ART model. Full wave numerical simulation results indicate that the system's thermal and viscous losses are governed by the geometry of the resonance unit cells. Together with the intrinsic trapping effect, this lossy ART metamaterials function as a high performance broadband sound absorber. It may contribute to the optimization of sound absorption materials and wedge design of anechoic chambers.

1:45

4pPAa4. Evaluation of the resolution of a metamaterial acoustic leaky wave antenna. Jeffrey S. Rogers, Christina J. Naify (Acoust. Div., Naval Res. Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375, jeff.rogers@nrl.navy.mil), Matthew Guild (U.S. Naval Res. Lab., National Res. Associateship Program, Washington, DC), Charles A. Rohde, and Gregory J. Orris (Acoust. Div., Naval Res. Lab, Washington, DC)

Acoustic antennas have long been utilized to directionally steer acoustic waves in both air and water. Typically these antennas are comprised of arrays of active acoustic elements which are electronically phased to steer the acoustic profile in the desired direction. A new technology, known as an acoustic leaky wave antenna, has recently been shown to achieve directional steering of acoustic waves using a single active transducer coupled to a transmission line passive aperture. The leaky wave antenna steers acoustic energy by preferential coupling to an input frequency and can be designed to steer from backfire to endfire, including broadside. This paper provides an analysis of resolution as a function of both input frequency and antenna length. Additionally, the resolution is compared to that achieved using an array of active acoustic elements. [This work was supported by ONR.]

2:00

4pPAa5. Holographic metamaterial using an acoustic leaky wave antenna. Christina J. Naify, Matthew D. Guild, Theodore P. Martin, Charles A. Rohde, David C. Calvo, and Gregory J. Orris (Acoust., Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil)

Acoustic antennas in the form of active phased arrays have long been utilized to steer acoustic waves in both air and water. Although acceptable, these antennas form electrically complex, bulky structures. Acoustic leaky wave antennas have recently emerged as a method to achieve directional steering using a minimal number of active transducers coupled to an analog metamaterial aperture. In this study, a two-dimensional air-acoustic leaky wave antenna, coupled to only four sources, is shown to steer acoustic energy within a full three-dimensional space while also providing significant savings in terms of size, weight, and cost. By careful selection of the analog geometry, the aperture has the capability to produce a wide range of holographic acoustic images.

2:15

4pPAa6. Electromechanical hybrid metamaterial for the control of ultrasonic guided waves. Nesrine Kherraz, Lionel Haumesser, Franck Levassort (GREMAN UMR CNRS 7347, Univ. François Rabelais of Tours, 03 Rue de la Chocolaterie, Blois 41000, France, nesrine.kherraz@univ-tours.fr), Paul Benard, and Bruno Morvan (Laboratoire Ondes et Milieux Complexes, UMR CNRS 6294, Univ. of Le Havre, Le Havre, France)

Last years, a growing number of research has been devoted to the control of the behavior of Phononic crystals and Metamaterials through for instance tunability of the frequency position and/or width of a band gap. This can be achieved by using active materials such as piezoelectric materials. Here, a new class of hybrid metamaterials (HMMs) is proposed to tune the dispersion of guided Lamb waves. The studied HMM is made of a single homogeneous piezoelectric plate on which metallic electrodes are laid on. These electrodes are very thin in comparison to the plate thickness and allow changing locally the electric boundary conditions (EBCs). It is shown, experimentally and theoretically, that by simply modifying the EBCs, coupling between some modes can be induced, leading to the opening of a gap at the edge or within the first Brillouin zone. Moreover, a hybridization gap is observed for the first symmetrical Lamb modes, involving the electrical resonance associated to an inductive electrical circuit connected on electrodes. Such a system opens important perspectives for the development of SAW devices for radiofrequency applications.

2:30

4pPAa7. Experimental determination of pressure-dependent stiffness of a nonlinear acoustic metamaterial. Stephanie G. Konarski, Michael R. Haberman, Preston S. Wilson (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu), Katia Bertoldi, Sahab Babaee (Harvard Univ., Cambridge, MA), and Jongmin Shim (Univ. at Buffalo, Buffalo, NY)

The field of acoustic metamaterials (AMM) involves the design of subwavelength structures to create material properties exceeding those of traditional composite materials. One such structure developed recently, known as a "Buckliball," displays strong pressure-dependent stiffness [Proc. Natl. Acad. Sci. USA **109**, 5978 (2012)]. The unique response of these structures results from element geometry: spherical elastomeric shells with thin, patterned circular membranes whose skeleton buckles when subjected to a differential pressure. We present measurements of the properties of a nonlinear effective medium consisting of Buckliballs immersed in water. The experimental apparatus consists of four elements: (1) a pressure vessel, which contains (2) a resonance tube filled with Buckliballs suspended in water, (3) an air-coupled ultrasonic system to monitor water height in the resonance tube, and (4) acoustic excitation and measurement instrumentation to obtain the frequency dependent acoustic response of the Buckliball and water-filled resonator. Estimates of the effective stiffness and density are obtained by measuring the fundamental resonance frequency and water column height as a function of overpressure. Experimental results are compared with model estimates and discussed in the context of nonlinear acoustic wave propagation. [Work partially supported by ONR.]

2:45

4pPAa8. Acoustic Scattering Cancellation: an Alternative to Coordinate Transformation Scattering Reduction. Theodore P. Martin, Charles A. Rohde (Code 7160, U.S. Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, theodore.martin@nrl.navy.mil), Matthew D. Guild, Christina J. Naify, David C. Calvo, and Gregory J. Orris (National Res. Council Res. Associate Program, Code 7160, U.S. Naval Res. Lab., Washington, DC)

Coordinate transformations have been extensively studied as a means to hide an object immersed in a fluid from an acoustic field. The coordinate transform approach relies on coatings with non-uniform and typically anisotropic material properties to guide an acoustic wave around the object. Acoustic scattering cancellation is an alternative approach that relies on modal cancellation within a desirable bandwidth to significantly reduce the acoustic scattering cross-section of an object. We present a direct comparison, using coatings with similar layer thicknesses and material properties, demonstrating that scattering cancellation achieves a similar level of scattering reduction compared to coordinate transforms, but with much thinner coatings. Furthermore, we present experimental results demonstrating significant reductions in the scattering cross-section of neutrally buoyant elastic cylinders suspended in an aqueous environment. Omnidirectional scattering reduction is obtained using a single isotropic coating through modal cancellation of the monopole and dipole modes of the cylinders. The modal cancellation is achieved by carefully tuning the coating's material properties and thickness such that the combined cylindrical object and coating exhibit effective medium properties identical to water. The scattering cancellation is close to 15 dB over a broad bandwidth for all reduced frequencies below $ka \cong 1$.

3:00–3:15 Break

3:15

4pPAa9. Iterative approaches to extend dilute homogenization predictions of Willis materials to higher volume fractions. Michael B. Muhlestein (Mech. Eng., Univ. of Texas at Austin, 3201 Duval Rd. #928, Austin, TX 78759, mimuhle@gmail.com) and Michael R. Haberman (Appl. Res. Labs. and Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Dynamic homogenization of generally heterogeneous elastic materials leads to three different effective material properties: stiffness, density, and parameters that couple momentum to strain and stress to velocity. This formulation was first proposed by Willis [Wave Mot. **3**, 1–11, (1981)] and is thus commonly referred to as Willis coupling. While there are some homogenization methods which account for Willis coupling, those models rely on either the assumption of periodic media or on dilute concentrations of coupled inhomogeneities in an otherwise homogeneous matrix. This work presents the use of dilute models with iterative micromechanical methods to predict general material properties in materials with no long-range order and elevated volume fractions of inhomogeneity. Two such methods, self-consistent and differential effective medium models, will be presented for the homogenization of composites of arbitrary anisotropy and Willis coupling resulting from the presence of Willis coupled inclusions in an elastic matrix material. Predictions from both models are presented and compared with similar predictions using homogenization schemes which do not account for Willis coupling. It is found that incorporating Willis coupling in these iterative approaches does not significantly affect the prediction of the tensors describing the effective stiffness and density of the mixture.

3:30

4pPAa10. Mie scattering obviates echoes in concert halls. James B. Lee (6016 S. E. Mitchell, Portland, OR 97206, cadwal@macforcego.com)

Mie-resonant-scattering from objects (of sizes similar to impinging waves) embedded in a plane surface has two aspects: it distributes sound into a pattern approximating a cosine law of diffuse reflection; it deconstructs phase relations among components comprising acoustic signals. The former is much better known than the latter. But in large rooms like concert halls, phase-preserving specular reflections constitute echoes, compromising musical information. Deconstructing phases of reflected musical signals by diffuse reflection confines temporal information to direct sound, presenting only the frequency spectrum in reflected sounds.

3:45

4pPAa11. Synthesis of wave surface by frame loudspeaker array with limited spatial frequency band. Akio Ando, Daisuke Yamabayashi, and Takuya Yamoto (Dept. of Electric and Electronics Eng., Faculty of Eng., Univ. of Toyama, 1-10-11 Kinuta Setagaya, Tokyo 157-8510, Japan, andio@eng.u-toyama.ac.jp)

We have studied a frame loudspeaker array for sound reproduction of three-dimensional television aiming at realizing clear localization and better sound depth control. Because such array has no secondary sources in its center region, the reproduced wave surface is sometimes deteriorated even if the least squares optimization will be used to reshape the surface. Over the past few decades, a considerable number of studies have been made on the sound field reproduction by loudspeaker array. Recently, higher order ambisonics (HOA) and near-field compensated HOA (NFC-HOA) attract attention of the researchers. It is well known that these methods can limit a

spatial frequency band by controlling the degree of spherical function. If such limitation is introduced to the frame loudspeaker array, the deterioration of wave surface will be alleviated because the optimization will be easy to progress by matching only the low spatial frequency components of the reproduced wave surface with those of the reference. The computer simulation showed that the least squares optimization with the moderate restriction of spatial frequency band brings better wave surfaces than that without the restriction.

4:00

4pPAa12. On the use of the Willis Constitutive Laws of Elastodynamics of Stratified Media in reflection/transmission problems. Olivier Poncelet and Alexander L. SHUVALOV (UMR CNRS I2M, Univ. of Bordeaux, I2M - Site Bat. A4, 351 Cours de la Liberation, TALENCE 33405, France, olivier.poncelet@u-bordeaux.fr)

The generalized constitutive relationships derived by Willis[1,2] for elastodynamics of composites have incited an increasing interest in the community of effective media, and particularly in that of metamaterials. The structures for which homogeneous and dispersive equivalent media are sought include laminates,[3,4] phononic crystals,[5] as well as locally resonant materials.[6] Those constitutive laws propose an extended vision of the notion of an effective medium since they fully generalize the linear relationship between the couple momentum/stress and the kinematic one particle velocity/strain through the tensors of anisotropic mass density (order 2), of elasticity (order 4) and of inertial coupling (order 3). Following general results obtained in [3] that provide the complete set of dispersive effective parameters describing exactly the "macroscopic" propagation in stratified media, this communication aims at exemplifying the use of the Willis model in different types of problems with interfaces coupling several media, at least one of which is actually inhomogeneous and described as a Willis effective medium. [1]G. W. Milton, J. R. Willis, Proc. R. Soc. A **463**, 855 (2007) [2]J. R. Willis, Mech. Materials **41**, 385 (2009) [3]A. L. Shuvalov *et al*., Proc. R. Soc. A **467**, 1749 (2011) [4]S. Nemat-Nasser and A. Srivastava, J. Mech. Phys. Solids **59**, 1953 (2011) [5]A. N. Norris *et al*., Proc. R. Soc. A **468**, 1629 (2012) [6]D. Torrent *et al*., Phys. Rev. B **92**, 174110 (2015).

4:15

4pPAa13. A novel image based approach to evaluate the room impulse response. Ambika Bhatta, Charles Thompson, and Kavitha Chandra (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu)

In this work, a novel exact image based solution for a room impulse response is formulated. The resulting response of the rectangular room is an exact solution which is represented in terms of image-sources and branch integrals. It is shown to resolve some of the numerical and analytical constraints of the image based, geometric acoustics models and mode based solutions. The results are quantified in terms of the characteristic parameters, frequency and the dimensions of the room. The computation is also shown by using the features of parallel programming.

THURSDAY AFTERNOON, 26 MAY 2016 — SALON H, 1:00 P.M. TO 3:45 P.M.

Session 4pPAb

Physical Acoustics: Multiple Scattering II

Valerie Pinfield, Cochair
Chemical Engineering Department, Loughborough University, UK, Loughborough LE11 3TU, United Kingdom

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

Contributed Papers

1:00

4pPAb1. Experimental validation of shear-mediated contributions to multiple scattering in concentrated random dispersions of spherical particles. Valerie J. Pinfield, Derek M. Forrester, and Jinrui Huang (Chemical Eng. Dept., Loughborough Univ., Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk)

Multiple scattering models for ultrasound propagation in dispersions of spherical particles have conventionally included only multiple scattering of the compressional wave mode. Recent developments to these models by Luppé *et al*. [J. Acoust. Soc. Am. **131**, 1113 (2012)] have incorporated the effects of mode conversions into the multiple scattering model; these arise from shear and thermal wave modes produced by scattering at each particle. In our recent work, we have reported the identification of the dominant contributions to effective attenuation in such dispersed systems for either solid or liquid particles, and have reported both analytical and numerical solutions for them. Here, we present the key results for *shear*-mediated multiple scattering effects which are dominant in concentrated systems of small *solid* particles (sub-micrometer) in the mega-Hertz frequency range. We show experimental validation of the model predictions for silica particles in the size range 100 nm to 1 micrometer and 1 to 20 MHz using two different spectroscopy techniques, first, a pseudo-continuous wave spectrometer (the Malvern Ultrasizer), and secondly a pseudo-random binary sequence cross-correlation spectrometer (Digusonic DSX) suitable for in-line process monitoring.

1:15

4pPAb2. Anomalous diffusion of multiply scattered ultrasound in a two-component disordered medium. Sébastien O. Kerhervé and John H. Page (Dept. of Phys. and Astronomy, Univ. of MB, Allen Bldg., Winnipeg, MB R3T 2N2, Canada, sebastien.kerherve@umanitoba.ca)

Measurements of multiply-scattered wave transport allow the characterization of heterogeneous media and may reveal anomalous wave properties. When a wave propagates through a sufficiently disordered medium, it will undergo many scattering events. This may lead to diffusive transport or

even, in highly disordered media, to localized behavior in which transport comes to a halt. These two regimes can be distinguished by measuring the evolution of the transverse intensity profile when point-like ultrasonic pulse is incident on the opposite side of a slab-shaped sample. In the case of diffusive propagation, the profile width grows without limit as the square root of time, while for localized waves, the width reaches a saturation value at long times due to the trapping of the waves in the medium. In our experiment, the sample consists of aluminum beads randomly packed in silicone oil. Contrary to expectations, the width goes through a maximum as a function of time and varies slowly afterwards. This novel behavior may be due to the existence of two coupled modes of propagation: a fast component travelling through the liquid and a slower component traveling through the bead network. The resonances of the beads influence strongly the propagation inside the solid network.

1:30

4pPAb3. Controlling the exceptional points of parity-time symmetric acoustics. Chengzhi Shi, Marc Dubois (Dept. of Mech. Eng., Univ. of California, Berkeley, 3112 Etcheverry Hall, Berkeley, Berkeley, CA 94720, chengzhi.shi@berkeley.edu), Yun Chen, Lei Cheng (Dept. of Microelectronics, Fudan Univ., Shanghai, China), Hamidreza Ramezani, Yuan Wang, and Xiang Zhang (Dept. of Mech. Eng., Univ. of California, Berkeley, Berkeley, CA)

Parity-time (PT) symmetric systems experience phase transition between PT "exact" and "broken" phases at exceptional point. These PT phase transitions contribute significantly to the design of single mode lasers, coherent perfect absorbers, isolators, diodes, etc. However, such exceptional points are extremely difficult to access in practice because of the dispersive behavior of most loss and gain materials required in PT symmetric systems. Here, we introduce a method to systematically tame these exceptional points and control PT phases. Our experimental demonstration hinges on an active acoustic element that realizes a complex-valued potential and simultaneously controls the multiple interference in the structure. The manipulation of exceptional points offers new routes to broaden applications for PT symmetric physics in acoustics, optics, microwaves, and electronics, which are essential for sensing, communication, and imaging.

1:45

4pPAb4. Acoustic metafluids with multiple negative-index bands. Thomas Brunet, Olivier Poncelet, Christophe Aristegui (UMR CNRS I2M, Univ. of Bordeaux, I2M - Site Bat. A4, 351 Cours de la Liberation, TALENCE 33405, France, olivier.poncelet@u-bordeaux.fr), Jacques LENG (UMR CNRS LOF, Univ. of Bordeaux, Pessac, France), and Olivier MONDAIN-MONVAL (UMR CNRS CRPP, Univ. of Bordeaux, Pessac, France)

The extraordinary properties of acoustic (random) metamaterials, such as negative refractive index, originate from low frequency resonances of sub-wavelength particles. While most of these functional materials are fabricated by mechanical engineering, we have recently shown that soft matter techniques coupled with microfluidics open a new synthesis route for acoustic metamaterials especially for ultrasonics [1]. As a demonstration, we have achieved 3D-bulk acoustic metafluids with an alternatively low, zero, and negative index by producing large amounts of calibrated soft porous microspheres, acting like strong Mie resonators [2]. The wide variety of physicochemical processes offered by chemical engineering allows for the tuning of the resonant particle properties over a broad range of mechanical/acoustical parameters. In this talk we show that, according to a fine control of the Poisson coefficient of the macro-porous resonators, it is not only possible to achieve soft acoustic metamaterial with one negative band, but also with two separate and tunable ones [3]. As both their width and depth depend on particle properties, the "soft approach" should benefit the design and fabrication of building-block composites with specific extreme properties required in some targeted applications, such as spatial control of sound for beamforming or cloaking. [1] Brunet *et al.*, *Science* **342**, 323–324 (2013). [2] Brunet *et al.*, *Nat. Mater.***14**, 384–388 (2015). [3] Raffy *et al.*, *Adv. Mater.* **DOI**: 10.1002/adma.201503524 (2016).

2:00

4pPAb5. Multi-layered elastic shell metamaterial elements for improved transmission in sonic crystals. Alexey S. Titovich (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., Bethesda, MD 20817, alexey.titovich@navy.mil), Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., NB, NJ), and Stephen D. O'Regan (Naval Surface Warfare Ctr., Carderock Div., Bethesda, MD)

A metamaterial-based sonic crystal, comprised of a lattice arrangement of cylindrical elastic shells, can be tuned to provide wave steering or phononic band filtering. The effective acoustic properties of such sonic crystals are determined by the geometric and material properties of each individual shell and the lattice spacing between shells. By matching the effective acoustic properties to those of the surrounding fluid, the sonic crystal can be made acoustically transparent at low frequencies, leading to improved transmission over that of non-matched shells. Bi-layered shells provide additional design parameters to broaden the region of acoustic transparency provided by a simple shell. In this effort, three-layered shells are employed to improve transmission further. Transformation acoustics analysis is applied to determine theoretical density and bulk modulus distributions, and then real-world materials are selected which best match these theoretical distributions. Finally, shell geometry is optimized to minimize scattering while constraining the effective acoustic properties to match those of the external fluid.

2:15

4pPAb6. Disk cavities in soft materials: Their bubble-like resonance and use in thin underwater sound blocking materials. David C. Calvo and Abel L. Thangawng (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, david.calvo@nrl.navy.mil)

The acoustics of gas-filled cavities in soft viscoelastic solids, such as rubber and gels, has been a renewed subject of research in recent years owing to usefulness in studies of multiple scattering and importance to sound insulating and anechoic materials. A brief review is followed by a presentation of recent research on disk cavity resonance done at the Naval Research Laboratory [J. Acoust. Soc. Am. **138**, 2537–2547]. A lumped parameter analysis of the breathing mode of a disk cavity is presented which yields a natural frequency expression valid for a high-aspect ratio cavity embedded in an elastic medium. A verification approach using finite-element methods is also described which directly computes resonance in the framework of COMSOL Multiphysics. Calculation of scattering cross sections and visualization of the elastic displacement field indicates the importance of shear wave radiation. As an application example, a specially designed single layer array of disk cavities in a thin silicone rubber (PDMS) sheet was modeled that resonantly blocks underwater sound by nearly 20 dB for a favorable wavelength/thickness ratio of 240. Disk cavities are found to provide a wider bandwidth than near-spherical cavities. [Work sponsored by the Office of Naval Research.]

2:30–2:45 Break

2:45

4pPAb7. An investigation of free flooding, air-filled underwater resonators. Andrew R. McNeese, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark S. Wochner (AdBm Technologies, Austin, TX)

This paper investigates the acoustic behavior of free flooding resonators consisting of inverted cups that are lowered into a water column such that a volume of air is entrapped within the cavity upon submersion. Similar to a two-fluid Helmholtz resonator, the resonance frequency is determined by the compliance of the entrapped air and the radiation mass of water at the opening of the cavity. The entrapped air mass is compressed as the resonator is lowered to depth due to the increase in hydrostatic pressure, which affects

the acoustic behavior. Collections of similar resonators with compliant walls were previously investigated for use in underwater noise abatement [Lee *et al*., Proc. Meeting Acoustics **22**, 045004 (2015)]; however, recent work has shown an increase in quality factor and attenuation performance for stiffer walled resonators. Measurements were taken to determine the resonance frequencies and quality factors of both individual resonators and arrays of resonators comprised of various wall materials as a function of water depth, resonator geometry, and spacing between resonators. Low frequency sound attenuation through arrays of resonators was also measured. These measurements and associated model comparisons will be discussed. [Work supported by AdBm Technologies.]

3:00

4pPAb8. Acoustic time reversal in granular media. Maxime Harazi, Yougu Yang, Mathias Fink, Arnaud Tourin, and Xiaoping Jia (ESPCI ParisTech, PSL Res. Univ., CNRS, 1 rue jussieu, Paris 75005, France, maxime.harazi@espci.fr)

In a non-dissipative medium, the wave equation is symmetric in time. Therefore, for every wave diverging from a pulsed source, there exists a wave that retraces all its original paths in a reverse order and converges at the original source. In the early nineties, M. Fink proposed a method for generating such a time-reversed wave. This method was first tested with ultrasound and then successfully extended to other types of waves such as microwaves, water waves, and even in optics. Several studies have shown that time reversal wave focusing is very robust to disorder. Here, we investigate time reversal (TR) of elastic waves propagating in fragile granular media consisting of glass beads under static compression. Pulsed elastic waves transmitted from a compression or a shear wave source are measured, time reversed, and back-propagated. The ability of the time-reversed wave to focus at the initial source is checked as a function of the source amplitude. We find that TR of the ballistic coherent wave is very robust to perturbations but provides poor resolution. By contrast, the short-wavelength scattered waves offer a finer TR focusing but are sensitive to rearrangements induced by the forward propagation wave itself: at large source amplitudes, time reversal focusing is broken, due to sound-induced rearrangements but without visible grain motion. Experimental results are confronted with predictions from a numerical model in which the propagation medium is modelled by a percolating network of masses interacting via linear springs.

3:15

4pPAb9. The emergence of spurious arrivals in Green's function extraction and passive imaging. Jie Li and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, San Diego, CA 92093, jil004@ucsd.edu)

Cross-correlations of ambient noise at two receivers can extract the two-point Green's function, given that the wave-field is spatially uniform. The presence of scatterers that act as uncorrelated secondary sources can destroy this condition. We retrieve the Green's function in one-sided noise with a scatterer in an aeroacoustic experiment, using a pair of microphones (separation 1 m) parallel to the beach which generate one-sided surf noise. The scatterer is a 20-cm-radius polyvinyl chloride pipe 2.5 m to the landside of the microphone pair. We cross correlate 400–2000 Hz noise to retrieve the Green's function. The results show that spurious scattered arrival emerges in the cross-correlation functions when the source distribution is limited. This causes the generalized optical theorem to break down, and thus, there is no guarantee that the spurious scattered arrival cancels. However, the spurious waves have a geometric interpretation, which are useful for inversion. By combining the travel times of the spurious and physical scattered waves, only a pair of receivers is sufficient to locate the scatterer passively.

3:30

4pPAb10. Reducing the pour point of crude oil by using of ultrasonic wave. Delong Xu, Weijun Lin, Jingjun Deng, Chao Li, Lixin Bai, and Yan Yang (Inst. of Acoust., Chinese Acad. of Science, Beijing 100000, China, xudelong@mail.ioa.ac.cn)

With the global economical development, high pour point crude oil (HPPCO) is getting more and more attentions. Due to its higher pour point and worse fluidity when it is cold, more difficulties are faced for its extraction. Ultrasonic wave technology is investigated to reduce the pour point of Sudan HPPCO in this paper. First, an ultrasonic horn is proposed and made. Then, after processing for 3 min by the horn, a decrease of more than 3 of the pour point of Sudan HPPCO is obtained. Furthermore, compared with that without ultrasonic processing, the percentage of higher molecular weight constituents becomes more, while that of heavier molecular weight component is less. The mechanism that the pour point of HPPCO can be changed through ultrasonic processing is analyzed and investigated by an experiment that three types of paraffin wax whose molecular weights are different are processed by ultrasonic wave. The main advantage of the ultrasonic processing for HPPCO is economical and environmental and the decrease of pour point after ultrasonic processing is irreversible.

Session 4pPP

Psychological and Physiological Acoustics and Speech Communication: Lessons from Interrupted Speech: Methods and Models

Valeriy Shafiro, Chair
Communication Disorders & Sciences, Rush University Medical Center, 600 S. Paulina Str., AAC 1012, Chicago, IL 60612

Chair's Introduction—1:30

Invited Papers

1:35

4pPP1. From macroscopic to microscopic glimpse-based models of intelligibility prediction. Martin Cooke (Ikerbasque, Lab de Fonetica, Facultad de Letras, UPV-EHU, Paseo de la Universidad, 5, Vitoria, Alava 01005, Spain, m.cooke@ikerbasque.org), Yan Tang (Univ. of Salford, Salford, United Kingdom), and Mate A. Toth (Univ. of the Basque Country, Vitoria, Spain)

Miller and Licklider's explorations of the intelligibility of temporally interrupted speech, and later studies extending their findings to the spectro-temporal plane, have shown how the twin factors of sparseness and redundancy confer a high degree of robustness on speech in noise. The current contribution addresses two questions. First, to what extent can quantitative estimates of supra-threshold unmasked speech account for average (macroscopic) intelligiblity across a range of speech styles and masking conditions? We examine how well glimpse-based objective intelligibility metrics predict listeners' speech recognition scores for natural and synthetic speech in the presence of stationary and fluctuating maskers, and demonstrate reduced correlations for competing sources with an informational masking component. The second question concerns which additional components, beyond speech glimpses, are required to make (microscopic) predictions of actual listener confusions at the level of individual noisy speech tokens. Using corpora of speech-in-noise misperceptions, we show that in many cases the source of listener confusions is the misallocation of information from the masker, suggesting that estimates of supra-threshold unmasked speech alone are insufficient to explain speech intelligibility in noise.

1:50

4pPP2. Recognition of interrupted words in isolation and in sentences. Gary R. Kidd and Larry E. Humes (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405, kidd@indiana.edu)

When listening to speech in the presence of competing sounds that render portions of a speech signal inaudible or unrecognizable, the ability to utilize partial speech information is crucial for speech understanding. To better understand this ability, we have been examining word recognition with different patterns of missing information (portions of speech replaced by silence) in words and sentences. Although we find that the major determinant of word recognition is the proportion of missing information, the location of glimpses and the pattern and predictability of glimpses can also influence performance. However, these effects depend on the presence or absence of a sentence context and the predictability of the target word within the sentence. With isolated words, the onset information is most important for recognition. For words in sentences, the importance of word onsets is diminished and depends on the amount of context provided by the sentence. When the pattern of interruptions throughout a sentence is manipulated, a predictable pattern of glimpses facilitates word recognition in a highly predictable sentence context, but not in a low-predictable context. The implications of these findings for theories of speech understanding under difficult listening conditions will be discussed. [Work supported by NIH (NIA and NIDCD).]

2:05

4pPP3. An information-theoretic approach to understanding interrupted speech. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Everyday listening conditions seldom present the listener with a completely intact speech signal. Research on understanding interrupted speech spans the last seven decades, yet few efforts have focused on what parts of the speech signal were being interrupted. Kewley-Port, Fogerty, and others explored the importance of consonants versus vowels for understanding speech when either was replaced by noise, but this approach is limited by their inherent differences such as segment duration. The present approach is inspired by Shannon information theory, where information is defined by unpredictability, uncertainty, or change. We developed a metric of biologically relevant spectral change in the speech signal, termed cochlea-scaled entropy (CSE). Sentences with high-CSE intervals (low predictability = high information) replaced by noise were understood more poorly than sentences with an equal number and duration of low-CSE intervals (high predictability = low information) replaced. CSE has been shown to better predict speech intelligibility than various temporal measures and consonant/vowel status. This approach has been validated across wide ranges of acoustic simulations of cochlear implant processing, suggesting that cochlear implant users might also utilize these information-bearing acoustic changes to understand speech. Extensions and future directions for this work will be discussed

2:20

4pPP4. Normal and Impaired hearing recognition of speech segments in noise. Jont B. Allen (ECE, Univ of IL, 1404 Sunny Acres Rd., Mahomet, IL 61853, jontallen@ieee.org) and Ali Abavisani (ECE, Univ. of IL, Urbana, IL)

The identification of very short phoneme segments of speech is the key to understanding speech in chopped noise. Over the last 12 years, UIUC has repeated Miller-Nicely's 1955 phone recognition experiment, with 60 subjects and six SNRs, from quiet to −22 dB SNR, providing an improve understanding of the fundamentals of phone perception in normal (NH) and impaired (HI) listeners. Our phone robustness metric (SNR50) is the SNR such that the phone error is 50% (Toscano and Allen (2014), JSHLR). The error rate at SNR50 + 5 [dB] is <0.33%. We interpret this to mean that above SNR50, phonemes are below the Shannon channel-capacity limit. This is a game-changer: We must reevaluate speech recognition methods. For example, in an experiment on HI ears, we found that HI ears make large errors (e.g., 100%) on a small subset of tokens (Trevino Allen, JASA **134**, 607, 2012). Averaging across tokens or listeners for any given consonant conflates the scores. There is a good news: Since there are only a small number of subject-dependent difficult sounds, testing time is reduced and accuracy is increased for a fixed test duration.

2:35

4pPP5. The contribution of amplitude modulations from speech and competing noise sources for speech recognition. Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1621 Greene St., Columbia, SC 29208, fogerty@sc.edu)

Speech is often heard in the presence of background noise during everyday listening conditions. Concurrent amplitude modulations of the speech and the interfering noise result in portions (i.e., "glimpses") of the speech signal that remain preserved at favorable signal-to-noise ratios while other portions of speech may be highly degraded. The acoustic and phonetic properties preserved during these glimpses play a role in determining the overall sentence intelligibility. Under such conditions, vocalic cues are relatively more preserved than consonantal/obstruent cues due to average differences in intensity. Investigations examining these different acoustic cues suggest important perceptual contributions from preserved speech amplitude modulations and non-simultaneous interactions from competing noise amplitude modulations. Results from several studies suggest that conditions that maximize the preservation of the sentence temporal envelope (e.g., during long glimpses or for vocalic intervals) result in higher levels of speech recognition. Amplitude modulation during noise-dominated intervals also impacts speech recognition performance. Overall, results suggest that the relative preservation and interaction (e.g., modulation masking or perceptual restoration) of amplitude modulations from the speech and noise is crucial for predicting speech recognition during noise interruption. [Work supported by NIH and ASHA.]

2:50–3:10 Break

3:10

4pPP6. Interrupted speech with competing talkers: Benefits of temporal envelope and periodicity cues for younger and older adults. William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, Medical Univ. of South Carolina, 135 Rutledge Ave., MSC 550, Charleston, SC 29412, bologna@musc.edu), Monita Chatterjee (Auditory Prostheses & Percept. Lab., Boys Town National Res. Hospital, Omaha, NE), and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Recognition of interrupted speech requires connecting speech fragments over time and across gaps of missing information. Intelligibility improves when silent intervals are filled with noise; the effect is enhanced when noise contains rudimentary speech information, such as the temporal envelope of the missing speech. In multiple-talker environments, recognition of interrupted speech is more difficult, particularly for older adults. In these cases, temporal envelope cues may provide an important scaffold for auditory object formation. Other basic speech cues, such as F0-related periodicity, may help listeners segregate multiple voices and provide additional benefit. The relative, and potentially additive, benefit of temporal envelope and periodicity cues, and their use by older adults, remain unclear. To address these questions, younger and older adults with normal hearing listened to sentences in quiet and competing talker backgrounds. Sentences were periodically interrupted with (1) silence, (2) envelope-modulated noise, (3) steady-state pulse trains, which contained periodicity information from the missing speech, or (4) envelope-modulated pulse trains, which provided both envelope and periodicity information. Results are discussed in terms of contributions of temporal envelope and periodicity cues to perceptual organization for younger and older adults listening in complex environments. [Work supported by NIH/NIDCD and a AAA Student Investigator Research Grant.]

3:25

4pPP7. Perceptual effects of fluctuating envelopes. Peggy B. Nelson (Univ. of Minnesota, 164 Pillsbury Dr. Se, Minneapolis, MN 55455, peggynelson@umn.edu) and Adam Svec (Starkey Hearing Technologies, Eden Prairie, MN)

Signal/masker envelope fluctuations have important effects on detection and discrimination. Narrowband Gaussian noise (GN) forward maskers yield higher masked thresholds for detecting pure tones than do low-fluctuation noise (LFN) forward maskers. The increased residual masking is due to inherent fluctuations in the temporal envelope of GN producing listener uncertainty. This uncertainty persists for longer durations for hearing-impaired (HI) than for normal-hearing (NH) listeners. In addition to listener uncertainty, amplitude-modulation (AM) forward masking may contribute to masking that occurs in complex listening tasks. In a recent study of AM forward masking, an unmodulated GN masker yielded more masking than an unmodulated LFN, suggesting that inherent envelope fluctuations were responsible for the amount of AM forward masking measured across listener groups. Contrary to predictions, there were no differences in AM forward masking between NH and HI listeners, revealing little effect of hearing loss on recovery from AM forward masking for this task. Considering the combination of listener uncertainty and AM forward masking, the persistence of masker envelope fluctuation effects likely lead HI listeners to experience sluggish recovery from prior rapid envelope fluctuations compared to NH listeners. Together, these findings may have implications for speech understanding in modulated or interrupted conditions. [Support: NIH-DC008306.]

3:40

4pPP8. Modulation masking and masked speech perception in normal-hearing school-age children and adults. Emily Buss, John H. Grose, Joseph W. Hall (UNC Chapel Hill, 170 Manning Dr., G190 Physicians, Chapel Hill, NC 27599, ebuss@med.unc.edu), and Christian Lorenzi (Ecole Normale Superieure & CNRS, Paris, France)

It has recently been suggested that adults' ability to recognize speech in noise is limited by the inherent modulation of that noise—a form of modulation masking. This study evaluated whether immature masked speech perception in school-age children could be due to greater susceptibility to modulation masking. To gauge modulation sensitivity, the first experiment measured sinusoidal modulation detection for rates of 10–300 Hz carried by a 5000-Hz pure tone. There were large individual differences, but little evidence of a child/adult difference. This adult-like modulation detection for a tonal carrier contrasts with published findings of adult/child differences in modulation detection for a noise-band carrier, suggesting that children may be more susceptible to modulation masking than adults. The second experiment evaluated masked sentence recognition for speech that was filtered into 28 adjacent equivalent rectangular bands (100–7800 Hz), with alternate bands presented to opposite ears. Maskers were composed of either noise bands or tones, one centered on each speech band. These stimuli have been argued to characterize effects of modulation masking. Young children tended to perform more poorly than adults overall. Masker effects will be discussed in terms of possible developmental differences in energetic and modulation masking.

3:55

4pPP9. Effects of adult aging on perception of alternated speech. Arthur Wingfield (Volen National Ctr. for Complex Systems, Brandeis Univ., MS 013, Waltham, MA 02453, wingfiel@brandeis.edu)

In 1954, Cherry and Taylor found that when presentation of a continuous speech message was rapidly alternated between the two ears, intelligibility progressively declined as the rate of alternation was increased up to 3–4 switching cycles/s (167–125 ms per ear). Intelligibility then improved as switching rates were increased beyond this point. This V-shaped function might occur if a finite time is required to switch attention from one ear to the other, during which time no usable information is available from either ear. Hence, the more frequently the speech is alternated between ears, the greater will be the cumulative loss of acoustic information. At alternation rates beyond 125 ms per ear the listener begins to adopt a strategy of attending to the interrupted signal from only one ear, relying on the redundancy of the large number of small speech segments to reconstruct the message (Miller and Licklider, 1950). We report data supporting the alternative view that the critical dimension underlying the point of minimal intelligibility is speech content per ear rather than time per ear. We then use this phenomenon as a test of whether adult aging results in a slowing of attentional shifts in audition.

4:10

4pPP10. Effect of language experience on the intelligibility of interrupted speech. Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu), Stanley Sheft (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX), and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL)

Signal distortions are more detrimental to speech perception by native than non-native listeners. We investigated how speech clarity and semantic context influence the perception of interrupted speech. Native and non-native American English listeners heard semantically meaningful or anomalous sentences produced as conversational or "clear" speech, gated at different rates (0.5–16 Hz). Results showed that both semantic context and speech clarity had a significant rate-dependent impact on the intelligibility of interrupted speech. In general, intelligibility was higher for native than non-native listeners. However, the magnitude of the clear-speech benefit varied across the two listener groups. The clear-speech benefit was obtained for gating rates of 2 Hz and above, except for meaningful sentences with native listeners where the benefit begin at 1 Hz. Both listener groups were able to use contextual information but native listeners derived more benefit at lower gating rates, indicating greater ability to access semantic information with limited acoustic-phonetic input. The results suggest that the non-native speech-perception deficit in adverse listening conditions may in part be due to their less efficient use of compensatory information at higher levels of language processing.

4:25–4:45 Break

4:45

4pPP11. Tales from the dip: Factors in cross-rate intelligibility variation of interrupted speech. Valeriy Shafiro, Stanley Sheft, and Brendan Prendergast (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu)

Speech intelligibility involves integration of temporally and spectrally distributed acoustic information into higher-order perceptual categories to obtain individual words. In 1950, Miller and Licklider pioneered a simple but powerful method of interrupting speech that has been extensively used to investigate factors that make speech signals perceptually robust. Among numerous subsequent studies, a consistent finding has been a nonmonotonic relationship between intelligibility and interruption rate. As interruption rate increases, and the duration of speech fragments in each interruption cycle decreases, a U-shaped rate-intelligibility function with a dip around 1–5 Hz frequently emerges. While many factors (e.g., speech materials, task parameters, listener age, or hearing status) have been shown to influence performance at specific interruption rates, reasons for the appearance and location of the dip in the function have remained obscure. Previous work indicates that the location of the dip in the rate-intelligibility function can be altered predictably with changes in the temporal structure of the interrupted speech stream, and may vary with the duration of the corresponding perceptual units. These findings will be considered in the context of neurophysiological and information-processing models of interrupted speech, and used to suggest a framework to guide future research and practical applications.

5:00

4pPP12. The intelligibility of interrupted, time-compressed speech. Michelle R. Molis and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov)

Speech is a highly redundant signal for young listeners with normal hearing, but access to that redundancy may be diminished for older listeners, with or without hearing loss. We compared the understanding of unprocessed speech with (1) time-compressed speech, (2) time-compressed speech expanded via interruptions, and (3) uncompressed speech interrupted with silence. Listeners were asked to identify the final four digits of spoken seven-digit strings presented in quiet and in a steady-state, speech-shaped background noise (SNR +5). Three uniform compression ratios (2:1, 3:1, and 5:1) were used. Participants were younger normally-hearing listeners (YNH), somewhat older listeners (mid-50s to mid-60s) well-matched to the YNH group in thresholds up to 4 kHz (ONH), and older listeners with moderate hearing impairment well-matched in age to the ONH group (OHI). Employing this method, we have demonstrated the relative importance of the amount of information contained between the interruptions; and, the influence of age and hearing loss on the effective use of that information. [Work supported by VA RR&D I01RX001020.]

5:15

4pPP13. Neuronal oscillations in decoding time-compressed speech. Oded Ghitza (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, oghitza@bu.edu)

At the core of oscillation-based models of speech perception is the notion that decoding is guided by parsing. In these models, parsing is executed by setting a time-varying, hierarchical window structure synchronized to the input. Syllabic parsing is into speech fragments that are multi-phone in duration, and it is realized by a theta oscillator capable of tracking the input syllabic rhythm, with the theta cycles aligned with intervocalic speech fragments termed theta-syllables. Prosodic parsing is into fragments that are multi-word in duration, and it is realized by a delta oscillator capable of tracking phrase-level prosodic information, with the delta cycles aligned with chunks. Intelligibility remains high as long as the oscillators are in sync with the input, and it sharply deteriorates once they are out of sync. In the pre-lexical layers, decoding is realized by a cascade of neuronal oscillators in the theta, beta, and gamma frequency bands, with theta as "master." This talk reviews a model that utilizes this cortical computation principle, capable of explaining counterintuitive data on the intelligibility of time-compressed speech hard to explain with conventional models of speech perception. [Work supported by AFOSR.]

5:30–5:50 Panel Discussion

Session 4pSC

Speech Communication and Signal Processing in Acoustics: Constraints, Strategies, and Economies of Effort in Speech Production: Joseph S. Perkell's Contribution to Speech Science

Jennell Vick, Cochair
Psychological Sciences, Case Western Reserve University, 11635 Euclid Ave., Cleveland, OH 44106

Frank Guenther, Cochair
Boston Univ., 677 Beacon Street, Boston, MA 02115

Lucie Menard, Cochair
Linguistics, Universite du Quebec a Montreal, CP 8888, Succ. Centre-Ville, Montreal, QC H3C 3P8, Canada

Chair's Introduction—1:30

Invited Papers

1:35

4pSC1. Sensory feedback control in speech: Neural circuits and individual differences. Frank H. Guenther (Boston Univ., 677 Beacon St., Boston, MA 02115, guenther@cns.bu.edu)

Speech production involves a combination of feedforward and sensory feedback-based control mechanisms. The latter have been characterized in experiments involving real-time perturbations during speech. Unexpected perturbations of auditory feedback result in corrective motor responses with a minimum latency of approximately 100 ms after perturbation onset. These responses have been shown for both pitch and formant frequency perturbations, and the responsible neural circuitry includes portions of the superior temporal gyrus and ventral premotor cortex (vPMC). Corrective responses to somatosensory perturbations (such as a downward force applied to the jaw) occur approximately 50 ms after perturbation onset and are mediated by ventral somatosensory cortex and vPMC. The degree to which an individual weights auditory versus somatosensory feedback varies substantially. Such differences rely in part on differences in sensory acuity, e.g., a speaker with relatively poor hearing is likely to rely more heavily on somatosensory feedback control mechanisms than auditory feedback control mechanisms. Additionally, reliance on feedback control may be modulated to compensate for impairments in the feedforward system for speech, e.g., adults who stutter show a reduced reliance on auditory feedback control compared to fluent speakers, perhaps because auditory feedback can have a deleterious effect on speech initiation in stuttering.

1:55

4pSC2. Sensorimotor development and speech production. Lucie Menard, Christine Turgeon, and Pamela Trudeau-Fisette (Linguist, Universite du PQ a Montreal, CP 8888, Succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

According to Perkell's view [J. Perkell, J. Neurol. **25**, 382–407], phonemic goals correspond to multidimensional spaces in the auditory and somatosensory dimensions. The relationships between sensory goals and motor actions in speech are acquired during infancy through feedback-based control mechanisms. Although there have been numerous studies of speech development, little is known about the specific roles of auditory and somatosensory feedback. This paper reviews a few studies of how pre-school-aged children rely on sensory feedback to reach phonemic targets. The studies looked at children with normal-hearing and deaf children with cochlear implants. Vowels and consonants were recorded in various prosodic contexts (neutral and under contrastive emphasis) and perturbation conditions (with or without artificial perturbations of the articulators). Acoustic, articulatory, and perceptual analyses were conducted to assess the degree to which children reached the targets. The data showed that although children's productions were more variable than adults, children generally adapted their articulatory gestures to produce the appropriate sensory speech goals. The effects of perceptual acuity and phonological knowledge in speech development are discussed.

2:15

4pSC3. Development of auditory and somatosensory goals for alveolar sibilant production in later childhood. Jennell C. Vick (Psychol. Sci., Case Western Reserve Univ., 11635 Euclid Ave., Cleveland, OH 44106, jennell@case.edu), Nolan T. Schreiber (Elec. Eng., Case Western Reserve Univ., Cleveland, OH), Rebecca L. Mental (Psychol. Sci., Case Western Reserve Univ., Cleveland, OH), Michelle L. Foye (Cleveland Hearing & Speech Ctr., Cleveland, OH), and Gregory S. Lee (Elec. Eng., Case Western Reserve Univ., Cleveland, OH)

An age-related increase in the fricative contrast /s/-/ʃ/, measured acoustically, occurs in children up to 7 years of age, with 5- and 7-year-olds producing contrasts that are greater than those of younger children, but still significantly smaller than the contrasts produced by adults. Kinematic studies have demonstrated that adult-like speech motor control does not emerge until later adolescence, although no articulatory studies of sibilant production in later childhood have been reported. In this study, the aim was to understand the relative use of somatosensory and acoustic/auditory goals for the production of the fricative contrast in older children (10–15 years of age) and adults. The fricative contrast was measured in the acoustic signal and in articulatory data gathered from the tongue blade. The acoustic and articulatory contrasts were analyzed to test the magnitude of the covariation of the two domains. We further analyzed the development of the contrast in both domains as a function of later speech development. Results will be discussed in the context of the hypothesis that sibilants are produced with prominent goals in both the somatosensory and auditory domains but that auditory goals predominate in older children because of continued refinement of feedforward commands.

2:35

4pSC4. Articulatory idiosyncrasy inferred from relative size and mobility of the tongue. Kiyoshi Honda (School of Comput. Sci. and Technol., Tianjin Univ., 135, Yaguan Rd., Jinnan Dist., Tianjin 300350, China, khonda@sannet.ne.jp), Honghao Bao, and Wenhuan Lu (School of Comput. Software, Tianjin Univ., Tianjin, China)

Origins of individual characteristics of speech sounds have been a mystery. Individual patterns of higher spectra could be attributed to quasi-static hypopharyngeal-cavity resonance, while those of lower spectra are puzzling because both spectra and vocal-tract shapes radically change during speech. A possible clue to look into articulatory idiosyncrasy may be the relation between relative size and mobility of the tongue in the oropharyngeal cavity. To this end, combined cine- and tagged-MRI collected from four Chinese speakers producing two-syllable words were processed. The relative tongue size was indexed by midsagittal tongue area divided by tongue plus airway area both measured above the level of the superior genial tubercle in static MRI during /i/. The mobility of the tongue was measured by average velocity of tag points located along the oral and pharyngeal surface of the tongue. In the result, the velocity monotonically decreased with the relative tongue size, suggesting that the smaller the tongue the faster the movement according to a speaker-specific anatomical constraint derived from the space available for tongue articulation in vowel production. [Work supported by National NSF Programs (No. 61573254 and 61304250), and National One-Thousand Program (WQ20111200010) in China.]

Contributed Paper

2:55

4pSC5. Patterns of lingual articulation: A real-time three-dimensional ultrasound + palate study. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu)

Joe Perkell's influence on the science of speech articulation has been enormous, beginning with the publication of his doctoral dissertation in 1969, which investigated the motion of speech articulators within the midsagittal plane. This presentation follows in the same spirit, extended to three dimensions. Three-dimensional (3D) tongue shapes—with aligned palate models—for a variety of speech sounds will be presented, followed by a principal components analysis of 3D tongue shape variability. The data were collected using a 3D/4D ultrasound system to image the tongue, and a 3D laser scanner to digitize palate impressions. Unlike most MRI data, these ultrasound recordings were made in upright sitting position during real-time speech. A limitation is the lack of information regarding the positions of the soft palate, the posterior pharyngeal wall, and the lips. The focus with 3D/4D ultrasound is therefore centered on the oral cavity and the anterior pharyngeal wall (i.e., the tongue root).

Invited Papers

3:10

4pSC6. Mind the gap: Electromagnetic articulometer observation of speech articulation in conversational turn-taking. Mark Tiede (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, tiede@haskins.yale.edu), Dolly Goldenberg (Linguist, Yale Univ., New Haven, CT), and Christine Mooshammer (Linguist, Humboldt-Universität, Berlin, Germany)

Joe Perkell pioneered the use of electromagnetic articulometry (EMA) for the observation and quantification of the kinematics of speech articulator movements. In the spirit of his research, we have extended these methods to EMA observation of two facing speakers interacting in conversation. The gaps or pauses between turns in speaking are known from acoustic measurement to be relatively short in duration, about 200 ms or the length of a syllable on average, and this gap duration has been shown to be consistent across widely diverse languages and cultures (Stivers *et al.*, 2009). However, because the cognitive latencies for producing a response are much longer than this interval its planning must occur during the incoming turn. Here we provide evidence for this planning from articulator movements that anticipate speech. Movements of sensors attached to the tongue, jaw and lips have been tracked for each of 12 speaker pairs. Gaps are measured as the difference between the acoustic end of one speaker's turn to the onset of aggregated articulator movement above a 20% peak velocity threshold for the respondent. Results show that the speech articulators typically assume an appropriate posture for initiation of a speaking turn well before the onset of speech.

3:30–3:50 Break

3:50

4pSC7. Speech biomechanics: What have we learned and modeled since Joseph Perkell's tongue model In 1974? Pascal Perrier (Gipsa-lab, CNRS & Université Grenoble Alpes, 11 Rue des Mathématiques, Saint Martin d'Hères F-38400, France, Pascal.Perrier@gipsa-lab.grenoble-inp.fr), Yohan Payan (TIMC-IMAG, CNRS & Université Grenoble Alpes, La Tronche, France), Mohammad A. Nazari (Mech. Eng., Univ. of Tehran, Tehran, Iran), Nicolas Hermant (TIMC-IMAG, CNRS & Université Grenoble Alpes, La Tronche, France), Pierre-Yves Rohan (Institut de Biomécanique Humaine Georges Charpak, Arts & Métiers ParisTech, Paris, France), Claudio Lobos (Departamento de Informatica, Universidad Tecnica Federico Santa Maria, Santiago, Chile), and Ahmad Bijar (TIMC-IMAG, CNRS & Université Grenoble Alpes, La Tronche, France)

With his "physiologically oriented, dynamic model of the tongue, Joseph Perkell introduced in 1974 a new methodological approach to understanding the "relationships among phonetic models and the properties and capabilities of the speech-production mechanism." This approach has guided a large part of our studies in the two last decades. In order to investigate how mechanical properties of the oro-facial motor system constrain the degrees of freedom of speech articulation and contribute to shaping the speech signals exchanged between speakers and listeners, we, among other research groups, have developed increasingly more realistic 2D, and then 3D, finite element (FE) biomechanical models of the human vocal tract and face. After summarizing some of our modeling and simulation results that shed light on some basic characteristics of speech production, we present recent developments which aim to improve the realism of the models: evaluation of the links between the FE mesh structure (based either on tetrahedra, hexahedra, or mixed elements) and simulation accuracy; development of an active 3D element that simulates muscle mechanics and muscle force generation mechanisms; use of Diffusion Tensor Imaging to investigate muscle anatomy; design of an Atlas-based method (i.e., without manual image segmentation) for the automatic generation of subject-specific models.

4:10

4pSC8. Acoustic and kinematic estimates of laryngeal stiffness. Victoria S. McKenna, Elizabeth S. Heller Murray, Yu-An S. Lien, and Cara E. Stepp (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cstepp@bu.edu)

Vocal fold movements differ between individuals with typical voices and those with voice disorders associated with increased laryngeal stiffness. Modeling suggests that changes in vocal fold kinematics correspond to changes in laryngeal muscle stiffness. In this study, 12 healthy adults produced repetitions of /ifi/ while varying their self-perceived vocal effort during simultaneous acoustic and nasal-endoscopic recordings, in order to compare a kinematic estimate of laryngeal stiffness to an acoustic measure. The acoustic measure, relative fundamental frequency (RFF), was determined from the last ten voicing cycles before the voiceless obstruent (offset) and the first ten voicing cycles of the following vowel (onset). A kinematic stiffness ratio was calculated by normalizing the maximum angular velocity by the maximum value of the glottic angle during vocal fold adductory gestures. A linear mixed-effect model found that RFF accounted for 52% of the variance in the kinematic data. Examined within-subject, 83% of participants exhibited at least a moderate negative linear relationship ($r = -0.5$ to -0.91) between the offset cycle 10 RFF and the kinematic stiffness ratio. Overall, the relationship between the acoustic and kinematic measures was strong and both measures showed consistent changes during self-modulated changes in vocal effort.

4:30

4pSC9. Speaking one's mind: Vocal biomarkers of depression and Parkinson disease. Satrajit S. Ghosh (McGovern Inst. for Brain Res., MIT, 43 Vassar St., 46-4033F, Cambridge, MA 02139, satra@mit.edu), Gregory Ciccarelli (MIT Lincoln Lab., Cambridge, MA), Thomas F. Quatieri (MIT Lincoln Lab., Lexington, MA), and Arno Klein (Sage Bionetworks, Seattle, WA)

Mental health disorders affect one in four adult Americans and have a staggering impact on the economy. Improved assessment and early detection of mental health state can reduce this impact. Voice analysis has been linked to depression and other mental health disorders. However, difficulties in data collection, variation in collection methods, and computational demands of analysis methods have limited the use of voice in mental health assessment. The pervasive use of smartphones offers a unique opportunity to understand mental health symptoms and speech variation in large groups. We have developed open source mobile applications to collect data and developed voice-based algorithms for predicting mental health state in major depressive disorder and Parkinson disease. We demonstrate the use of biophysical speech production models in creating and improving features for machine learning, in contrast to traditional approaches for feature extraction. A model-based approach fuses prior knowledge of the system with the input data, constrains the space of parameters to biophysically realistic values, and reduces overall prediction error. This joint work with MIT Lincoln Laboratory and Sage Bionetworks couples mobile sensors to effective feature extraction and prediction models to enable a scalable approach for estimating individual variation in mental health disorders.

4:50

4pSC10. Objective assessment of vocal hyperfunction. Robert E. Hillman, Daryush Mehta (Voice Ctr., Massachusetts General Hospital, One Bowdoin Square, Boston, MA 02114, hillman.robert@mgh.harvard.edu), Cara Stepp (Boston Univ., Boston, MA), Jarrad Van Stan (Voice Ctr., Massachusetts General Hospital, Boston, MA), and Matias Zanartu (Universidad Tecnica Federico Santa Marıa, Valparaiso, Chile)

Vocal hyperfunction (VH) is associated with the most frequently occurring types of voice disorders including benign vocal fold lesions (e.g., nodules) and dysphonia that occurs in the absence of concurrent pathology (e.g., muscle tension dysphonia). In 1989, Hillman *et al*. (J. Speech Hear. Res. **32**, 373–392, 1989) proposed and initially validated the first experimental framework for hyperfunctional voice disorders that was based on using objective measures of vocal function to test basic concepts about VH. This initial work set the stage for a program of research aimed at improving the prevention, diagnosis, and treatment of hyperfunctional voice disorders by attaining a better understanding of the etiological and pathophysiological mechanisms that underlie specific disorders within the broad

4p THU. PM

range of those associated with VH. An update on the progress of this ongoing work will be provided including new insights gained through the use of ambulatory voice monitoring, machine learning, and computer modeling of phonatory mechanisms. [Work supported by NIH-NIDCD R33 DC011588 and the Voice Health Institute.]

5:10

4pSC11. Collaboration and results in research on speech motor control. Joseph S. Perkell (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA 02139, perkell@mit.edu)

Research on speech motor control was conducted in the Speech Communication Group in R.L.E. at M.I.T. between 1965 and 2012. Virtually all of this work was collaborative—with participation of group members and colleagues from other institutions in the United States and abroad. The work dealt with various aspects of speech production, including properties of the production and perception mechanisms and their influences on observed behaviors and control strategies. Examples will be presented that illustrate the indispensable roles of collaborators in studies of motor equivalence, relations between production and perception and the roles of feedback and feedforward control. Collaborators' contributions were based on their expertise in a number of areas, including: electrical and biomedical engineering, experimental psychology, acoustic phonetics, speech perception, statistical analysis, neuro-computational modeling, and design of sophisticated hardware and software. These examples demonstrate how such collaborations were essential to a productive output of research findings and theoretical advances for over four decades. [Research supported by NINCDS and NIDCD, NIH.]

THURSDAY AFTERNOON, 26 MAY 2016 SNOWBIRD/BRIGHTON, 1:30 P.M. TO 5:05 P.M.

Session 4pSP

Signal Processing in Acoustics and Underwater Acoustics: Detection and Estimation in Uncertain Acoustic Environments II

Paul J. Gendron, Chair
ECE Department, University of Massachusetts Dartmouth, 285 Old Westport Road, North Dartmouth, MA 02747

Invited Papers

1:30

4pSP1. Bayesian model selection for a broadband coprime array with unknown number of broadband sources. Dane R. Bush and Ning Xiang (Architectural Acoust., Rensselaer Polytechnic Inst., 2609 15th St., Troy, NY 12180, danebush@gmail.com)

Coprime microphone arrays use sparse sensing to achieve O(MN) degrees of freedom using only O(M + N) elements, where M and N are coprime integers. The benefit is a narrow beam at frequencies higher than the spatial Nyquist limit allows, with residual side lobes arising from aliasing. These side lobes can be mitigated when observing broadband sources [D. Bush and N. Xiang, *J. Acoust. Soc. Am.,* **138**, 447–456 (**2015**)]. Peak positions indicate directions of arrival in this case; however, uncertainties on number of concurrent sound sources in practical applications challenge classical approaches to direction-of-arrival estimations. One has to first resolve the uncertainty on how many sources are present. In this work, Bayesian inference is used to first select which model the data prefer from competing models before estimating model parameters, including the particular directions of arrival. The model is a linear combination of modified Laplacian distributions (one per sound source). The posterior probability function is explored over the entire parameter space by nested sampling in order to evaluate Bayesian evidence for each model. Bayesian evidence is crucial for resolving the uncertainties regarding number of sources, preferring simpler models while penalizing unnecessarily complicated models in an inherent implementation of Occam's razor.

1:50

4pSP2. Bayesian localization, tracking, and environmental inference. Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper presents an overview of a general Bayesian inference approach to source localization, tracking and/or environmental estimation. Source location and spectral parameters together with environmental parameters are all considered unknown random variables to be estimated from prior information and observed acoustic data. The relative level of prior information for various parameters differentiates applications of interest. For instance, controlled-source geoacoustic inversion typically involves large prior uncertainties for seabed parameters but small uncertainties for source locations, although some applications, such as inverting noise from ships-of-opportunity, may involve larger location uncertainties. Alternatively, source localization in an uncertain environment typically involves non-informative location priors and environmental priors that reflect available knowledge. Tracking applications include additional prior constraints on source speed. In all cases, the goal is to compute marginals of the posterior probability density for source and environmental parameters, quantifying the information content of the data and prior. This is typically carried out with Markov-chain Monte Carlo methods including Metropolis-Hastings sampling and/or Gibbs sampling, with various approaches applied to improve efficiency (e.g., principal-component sampling, parallel tempering) and generality (trans-dimensional inversion). Multiple-source localization minimizes the Bayesian information criterion to estimate the number of sources.

2:10

4pSP3. Broadband cross-correlation processing—Taking advantage of high-frequency impulse response envelope at low frequency. Paul Hursky (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

The impulse response in a refracting ocean with multipath can readily be modeled using ray tracing (using a high-frequency approximation) or measured directly using acomms systems (using high-frequency broadband waveforms). Forming the envelope of this impulse response results in a robust characterization of the channel, based on time differences and angles of arrival, that are well-resolved by virtue of the bandwidth and short wavelength at high frequency. Can such a characterization be used as an aid to processing lower frequency signatures that are narrower in bandwidth, and thus not as well resolved in angle, and more prone to fading due to multi-path mutual interference? Paths that interact with a layered seabed will introduce frequency-dependent effects that must be accounted for. We will present results of processing data from several vertical line receive arrays, where we have a variety of broadband and narrowband waveforms being transmitted from several towed sources, and can leverage the high-frequency transmissions as an aid to processing the low-frequency signatures.

2:30

4pSP4. Source symbol decisions in the presence of space and time varying shallow water acoustic response functions. Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, pgendron@umassd.edu), Kari Cannon, and Graham Entwistle (ECE Dept., Univ. of Massachusetts Dartmouth, Darmouth, MA)

Effective underwater acoustic communications requires source symbol decisions in the presence of an uncertain space and time varying acoustic response function. A hierarchical mixture Gaussian model is useful for modeling both the sparsity of arrivals as well as their spread in angle, Doppler and propagation delay [Canadian Acoust. **40**]. In this framework, the delay spread, the degree of sparsity, and the Doppler spread all must be marginalized based on the observed data. We discuss the degree of sparsity as well as the correlation among multi-path arrival times and how these uncertain features in the response function can be efficiently treated in a computationally reasonable and statistically efficient manner via the hierarchy. The approach relies on iterative coherent symbol decisions with an empirical Bayes approach to estimating the hyper-parameters of the model permiting flexibility to adapt to environmental conditions. It is shown that coherent multi-path combining and Doppler compensation are possible at extremely low signal to noise ratios (i.e., < -18 dB), at ranges in excess of 1 km and with throughputs exceeding 100 bps with single element reception. Results are shown for large bandwidth M-ary orthogonal sequences tailored only to a maximum allowable multipath spread.

Contributed Papers

2:50

4pSP5. Adaptive pulse compression for clutter mitigation in midfrequency active sonar. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com), Jonthan Botts (Appl. Res. in Acoust. LLC, Culpeper, VA), Charles F. Gaumond (Appl. Res. in Acoust. LLC, Washington, DC), and Ian Cummings (Appl. Res. in Acoust. LLC, Culpeper, VA)

Midfrequency active sonar can achieve a combination of fine range resolution and good signal-to-noise ratio (SNR) by transmitting low-crest-factor frequency-modulated (FM) waveforms that are pulse compressed by match filtering. While this linear filter optimizes SNR for signals in additive Gaussian noise, it has large-amplitude range sidelobes that can allow strong sources of clutter, such as fish schools, to mask weaker nearby targets. To address range sidelobes we consider the reiterative minimum mean-square error (RMMSE) adaptive-pulse-compression algorithm [Blunt and Gerlach, *Proc. IEEE Intl. Radar Conf.*, Sept. 2003]. RMMSE performance is known to degrade for small Doppler shifts in the received waveform (e.g., due to clutter internal motion), which, depending on the form of the FM sweep, can be partially mitigated by covariance-matrix tapers [Cuprak, M.S. Thesis, George Mason University, 2013]. We note that RMMSE is analogous to the minimum-variance distortionless-response (MVDR) beamformer: for each range cell it steers nulls the location of strong returns at nearby range samples. Motivated by this similarity and expectation of target sparsity in range, we extend the compressive beamforming approach developed in prior work to learning a compressive-sensing match filter. Preliminary results from this work are discussed. [Work supported by a NAVSEA Phase I SBIR award.]

3:05–3:20 Break

3:20

4pSP6. Inferences on target speed, depth, and range from a continuous wave transmission. Paul J. Gendron, Tamunoala Charles-Ogan (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, togan@umassd.edu), David C. Anchieta (ECE, Universidade Federal do Pará, Belém, Brazil), and Justin Conor (ECE Dept., Univ. of Massachusetts Dartmouth, Darmouth, MA)

Tracking an underwater mobile object by means of a continuous wave transmission is challenging in part due to the difficulty of drawing inferences on closely spaced tones associated with target depth and speed. Considered here is a bistatic sonar arrangement employed to infer the depth, speed and range of an oncoming submerged object. Computation of the full posterior probability distribution of the returned amplitudes and frequencies from both the prior distribution and a finite duration window of the received waveform is made by Markov-chain Monte Carlo sampling. A Gibbs sampler is employed to construct the posterior joint density of all parameters by taking full advantage of the analytic tractability of the conditional and marginal densities of the received amplitudes while those of the ordered frequencies are constructed numerically by either an inverse quantile sampling or a Metropolis-Hastings sampler. The inferred density of depth, range, and speed of the target is accomplished by constructing a numerical inverse-transformation of the forward propagation model.

3:35

4pSP7. Target localization in a reverberant shallow ocean waveguide with environmental uncertainty using a nonlinear frequency-difference signal processing technique. Brian M. Worthmann and David R. Dowling (Univ. of Michigan, 1231 Beal Ave., 2010 Lay Automotive Lab., Ann Arbor, MI 48109, bworthma@umich.edu)

Model-based signal processing for active sonar localization is typically infeasible due to insufficient knowledge of the acoustic environment. Additionally, in a shallow ocean environment, surface or bottom roughness create diffuse reverberation that can often obscure a desired target echo. Recently, a nonlinear signal processing technique was developed for passive acoustic source localization (*Worthmann, et al.,* 2015) that is applicable to high frequency sources in uncertain shallow ocean environments. This technique exploits a nonlinear field product (the autoproduct) and uses in-band hydrophone array measurements to determine field information in a lower, out-of-band, frequency regime where environmental uncertainties are less detrimental. When extended to monostatic active sonar with a vertical array, this technique allows a model-based signal processing algorithm to combat the detrimental effects of reverberation. The nonlinear signal processing algorithm is presented, along with simulations in a 5-km range, 200-m deep ideal waveguide with environmental uncertainties and significant reverberation at frequencies between 2- and 5-kHz. Successful detection and localization of a mid-water-column target is found to be possible at simulated signal-to-reverberation levels as low as -5 dB. Comparisons to existing signal processing detection and localization algorithms are provided. [Sponsored by the Office of Naval Research and the National Science Foundation.]

3:50

4pSP8. Detection of an object bottoming at seabed through modeling consecutive reflected signals. Sunho Kim, Sungbin Im (School of Electron. Engr., Soongsil Univ., 369, Sangdo-Ro, Dongjak-Gu, Seoul 06978, South Korea, sbi@ssu.ac.kr), Taebo Shim, and Steve G. Kargl (APL-UW, Seattle, WA)

In various fields, it is very important to detect an object bottoming at seabed. In this paper, a scheme is proposed for detecting an object bottoming at seabed in the shallow water. In this work, a monostatic active sonar system is considered to use a linear frequency modulated signal as a transmitted signal, and to send consecutive pings in the shallow water. An object is assumed to be randomly located within 1000 m from the transmitter. A reflected signal received at the receiver can be modeled using a previously received reflected signal with Wiener filtering, to produce a FIR filter with estimated coefficients. This filter can be applied to predict the next received signal with the current received signal as input. The energy can be computed from the error signal between the predicted and received signals. In the case when the object is not located in the search area, the reflected signal is very similar to each other, and thus the energies of the error signals are very small. However, in the case when the object is placed and the reflected signal is scattered from the object, the energy of the error signal may be increased because the coherence between the predicted and received signals is low. The advantage of this approach is not to require a prior information on the characteristics of the sediment with the assumption that the seabed is consisting of an uniform sediment. The verification of the proposed scheme is investigated in terms of detection probability and detection range through computer simulation.

4:05

4pSP9. Guided wave reconstruction in complex geometries with a dictionary learning framework. K. Supreet Alguri and Joel B. Harley (Elec. and Comput. Eng., Univ. of Utah, 40 South 900 East, apt 2G, Salt Lake City, UT 84102, kssupreet@gmail.com)

Guided waves are an attractive tool for structural health monitoring (SHM) due to their ability to interrogate large areas of a structure. Yet, guided waves are characterized by multi-modal and frequency dispersive behavior and quickly grow in complexity with the structure. For example, guided wave reflections from plate edges, fasteners, or joints will lead to complex, difficult to analyze data. Most SHM algorithms try to remove or ignore these reflections. Yet, knowledge about the reflections and their acoustic behavior can significantly improve detection and localization algorithms. Hence, accurate knowledge about guided waves reflections is of a significant interest in SHM. In this paper, we reconstruct and predict guided wave measurements from geometric environments with reflections. We leverage dictionary learning and sparse recovery algorithms to achieve this goal. Dictionary learning is used to learn the building blocks of guided waves from simulation data. Sparse recovery algorithms are used to create predictive models of wave propagation based on experimental data. From simulation results, we show that our framework can successfully predict wave propagation, including reflections, across an aluminum plate with an accuracy of more than 95% from just 20 measurements. We demonstrate similar results with experimental data.

4:20

4pSP10. Design and tests of an acoustoelectric logging tool. Junqiang Lu, Xiaodong JU, Honglin Zhao, Baiyong Men, Wenxing Duan, and Wenxiao Qiao (China Univ. of Petroleum-Beijing, 18# Fuxue Rd. Changping District, Beijing 102249, China, lujq@cup.edu.cn)

The acoustoelectric logging is a developing method, and the logging method can interpret the seepage properties of the pore formation especially, such as permeability and so on. A new acoustoelectric logging tool is presented. The tool is composed of two acoustic transmitting transducers, three acoustic receiving transducers, two excitation electrodes, and four receiving electrodes. The transmitting transducers can work in a phased mode, and more radiant energy can be effectively generated. The acoustic radiant energy is a very important factor for the acoustoelectric logging tool, and measurements are conducted in an anechoic tank with dimensions of 5.0 m $\times$ 5.0 m $\times$ 4.0 m. The pulse width of excitation signals is $70\mu s$, and the peak value is above 3000 V. The main frequency of receiving waves is 9.52 kHz, sound pressure is 47.2 kPa, and transmitting voltage response level is 147.5 dB. The tool is tested in the experimental wells and exploration wells. While the acoustic transmitting transducers radiate acoustic excitation signals, acoustoelectric signals from interfaces and concomitant acoustoelectric signals with acoustic waves are all acquired in clastic rock formations.

4:35

4pSP11. Bayes Factor test for the discrimination of a submerged mobile object from a continuous wave transmission. Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, North Dartmouth, MA), David C. Anchieta (ECE, Universidade Federal do Pará, Universidade Federal do Pará, Belém, Brazil, davidca102@gmail.com), Tamunoala Charles-Ogan, and Graham Entwistle (ECE Dept., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Detection of an underwater mobile object by means of a continuous wave transmission is a challenging problem in part due to the difficulty of making inferences on closely spaced tones that provide clues regarding the objects' depth and speed. Considered here is a bistatic sonar arrangement to detect the presence of an oncoming submerged object. The optimal receiver computes the Bayes factor associated with the hypothesis test. The analytic intractability of the marginalization associated with the composite nature of the hypothesis leads to numerical methods of integration. The prior density on the target present scenario is constructed by an inverse image transformation through the forward propagation model. Computation of the Bayes factor is accomplished by Markov-chain Monte Carlo sampling with a Gibbs sampler. Analytically tractable conditional and marginal densities of the tone amplitudes are exploited while the conditional density of the ordered frequencies are constructed numerically by an inverse quantile sampler. Detection performance of a receiver against a mobile target are illustrated and discussed.

4:50

4pSP12. Merging models and data: Predictive modeling for guided wave. Joel B. Harley, K. Supreet Alguri (Dept. of Elec. and Comput. Eng., Univ. of Utah, 50 S Central Campus Dr. Rm. 3104, Salt Lake Cty, UT 84112-9249, joel.harley@utah.edu), and Alexander Douglass (Dept. of Mech. Eng., Univ. of Utah, Salt Lake Cty, UT)

New simulation tools for nondestructive evaluation (NDE) and structural health monitoring (SHM) are enabling predictive power that will quicken inspection accuracies, minimize inspection costs, reduce the need for data, and create pathways to new automatous inspection and monitoring methods. Simulations can be especially powerful tools for analyzing

complex geometries and modern anisotropic materials, such as carbon-fiber reinforced composites, where NDE and SHM theory and practice is still developing. Yet, truly predictive simulations are yet to be realized. Most simulations rarely match with experimental data unless the simulation is meticulously tuned. In this paper, we describe a framework for merging existing models with experimental data to create better predictive simulations of guided waves and other complex acoustic environments. We leverage tools and theory from compressive sensing, sparse inversion, and convex optimization. We focus on guided waves due to their significant complexity and their wide use in SHM and other acoustic applications. Our results achieve prediction accuracies of greater than 90% for guided waves in both isotropic and anisotropic environments. We also demonstrate how predictive models can be used for a variety of applications, including time of arrival estimation and temperature compensation.

THURSDAY AFTERNOON, 26 MAY 2016 SALON A, 1:30 P.M. TO 4:05 P.M.

Session 4pUW

Underwater Acoustics: Acoustic Propagation in the Ocean

Ying-Tsong Lin, Chair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543

Chair's Introduction—1:30

Contributed Papers

1:35

4pUW1. Practical estimates of the coherent leakage of sound from a mixed layer surface duct. Adrian D. Jones (Maritime Div., Defence Sci. and Technol. Group, P.O. Box 1500, Edinburgh, SA 5111, Australia, bear-jones@adam.com.au), Alec J. Duncan (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, WA, Australia), and Zhi Y. Zhang (Maritime Div., Defence Sci. and Technol. Group, Edinburgh, SA, Australia)

In a mixed-layer surface duct, acoustic transmission at frequencies below that required for duct trapping, as well as at frequencies moderately higher, is subject to coherent leakage of energy to the thermocline. The rate of coherent leakage with range is related to the imaginary part of the horizontal wavenumber of a mode, for which an iterative technique is commonly used for evaluations. In order to obtain alternative direct analytic expressions of the leakage rate, the analysis of Furry, described, for example, by Pederson and Gordon (JASA **47**, 304–326, 1970), has been revisited. Consideration has been given to both very large and very small ratios between sound speed gradients in the duct and below the duct, and ratios of sound speed gradients typical of those encountered as sea. The leakage rate for the first mode was found to well approximate the total signal leakage, hence the proposed analytical expressions relate to the first mode. Leakage values based on these expressions are compared against results from both a modal model and a wavenumber integration model, and a pre-existing approximate expression for leakage, for a number of surface ducted scenarios for frequencies relevant to the onset of duct trapping.

1:50

4pUW2. Acoustic influences of front width in a coastal curved-front model. Brendan J. DeCourcy (Mathematical Sci., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, decoub@rpi.edu), Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Analysis of an ocean shelf-slope front model [Y.-T. Lin and J. F. Lynch, J. Acoust. Soc. Am. **131**, EL1–EL7 (2012)] shows how acoustic quantities such as horizontal wavenumber depend on feature parameters [DeCourcy *et al*., J. Acoust. Soc. Am. **137**, 2421 (2015)]. The front model has a sharp curved interface between two isospeed regions in a coastal wedge. In this talk, the front is modeled more realistically with continuous sound speed variation over an interval of specified width between the two isospeed regions. The same approach is used as for the sharp front to obtain expressions for acoustic normal modes in the wedge. The solution comparisons between the sharp and continuous fronts are described, including the characterizations of modes as trapped, leaky, or transition. Simplified forms of the dispersion relation for the horizontal wavenumber eigenvalues are investigated. The objectives are to examine how including a continuous sound speed variation changes any conclusions about parameter dependence for the sharp front, and to determine sensitivity to model parameters such as front location, front width, slope angle, source frequency, and sound speed. [Work supported by the ONR.]

2:05

4pUW3. Coupled acoustic mode equations in the ocean waveguide with rough sea surface and elastic bottom. Andrey K. Morozov (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com) and John A. Colosi (Dept. of Oceanogr. Graduate School of Eng. and Appl. Sci., Naval Postgrad. School, Monterey, CA)

Sound wave scattering caused by rough surfaces has been the subject of investigation for many decades. The problem has no exact analytic solution and approximate approaches are needed. In the ocean, scattering of sound by perturbed ocean boundaries couples acoustic modes and increases attenuation. In this paper, approximate coupled mode equations are analyzed using a small perturbation approach. The analytic solution connecting mode shapes with coupling mode coefficients has been derived. It is shown that the coupled mode matrix can be approximated by a linear function of one parameter, which is the perturbation of the sea surface. The equations are applied for one-direction sound propagation with a rough sea surface. For this one parameter model, the coupled mode differential equation has an easy solution in the form of a matrix exponent and can be useful for many applications. The proposed method can be easily extended to the case where modes are coupling by uneven bottom topography including elastic properties for compressional and shear waves.

2:20

4pUW4. Examination of rough surface scattering from water/air interface using hybrid parabolic equation model. Mustafa Aslan (Dept. of Phys., Turkish Naval Acad., 1 University Circle, Monterey, CA 93943, maslan1@nps.edu) and Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA)

Many ocean acoustic propagation models assume an idealized pressure-release boundary at the ocean surface. This is easily accomplished numerically using finite element, finite difference, and split-step Fourier techniques. For models based on the split-step Fourier algorithm, rough surfaces can also be treated, but require additional complexity in the definitions of the field and propagator functions through a field transformation technique. For this reason, it may be advantageous to model a rough water/air interface in a manner analogous to the bottom treatment by extending the calculation into the air medium, thereby simplifying the definitions of the propagator functions. However, standard approaches to treat density discontinuities in split-step Fourier algorithms invoke smoothing functions, which have been shown to introduce phase errors in range. In this work, the hybrid split-step/finite-difference approach introduced by Yevick and Thomson (1996) is implemented in the Monterey-Miami Parabolic Equation (MMPE) model for both the water/sediment and water/air interfaces. Particular attention is paid to comparisons between the rough surface scattering results from the field transformation technique (applied to a pressure release surface) and the hybrid split-step/finite-difference approach (applied to the water/air interface).

2:35

4pUW5. Sound propagation from the Canadian Basin to the Chukchi shelf during Summer 2015. Mohsen Badiey, Justin Eickmeier, Andreas Muenchow (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu), Y T. Lin (Woods Hole Oceanographic Inst., Woods Hole, Delaware), Timothy Duda, John Kemp (Woods Hole Oceanographic Inst., Woods Hole, MA), Matthew Dzieciuch, and Peter Worcester (Scripps Inst. of Oceanogr., La Jolla, CA)

During the summer of 2015 a pilot experiment for the Canadian Basin Acoustic Propagation Experiment (CANAPE) was conducted between deep Arctic basin and shallow Chukchi shelf. A vertical line array (VLA) was deployed on the shelf (72.336 N, 157.449 W) at a depth of 161 m from July 26 to August 13, 2015. Sound sources were deployed from R/V Sikuliaq at locations ranging from 131 to 375 km from the VLA. M-Sequences centered at 250 Hz (bandwidth of 62.5 Hz) were transmitted from each location. Discrete shipboard CTD data were conducted at each transmission location and continuous CTD data were recorded along the VLA. Water column data suggests a sound channel located vertically between Pacific and Atlantic waters near the depth of Arctic halocline layer. They also show upward shoaling of lower halocline waters onto the shelf. This upwelling over the continental slope moves the sound channel from offshore to onshore near the bottom. The axis of this sound duct is 100 m below the surface and is about 100 m thick. Received acoustic signals on the shelf increase rapidly in intensity at time scales that range from minutes to days. In this paper we show the acoustic variations are due to the variable sound speed profile. Signal intensity, noise, and temporal variability for geotime scale of minutes to hours are reported. Numerical simulations are conducted to investigate the associated acoustic effects in data. [Work supported by ONR 322OA.]

2:50–3:05 Break

3:05

4pUW6. Numerical analysis of underwater sound propagation over the Chukchi Sea shelfbreak. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu), Mohsen Badiey (Univ. of Delaware, Newark, DE), Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Andreas Muenchow (Univ. of Delaware, Newark, DE), John Kemp (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Justin Eickmeier (Univ. of Delaware, Newark, DE), Matthew Dzieciuch, and Peter Worcester (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA)

During the 2015 Canada Basin Acoustic Propagation Experiment (CANAPE) expedition on the southern edge of Canada basin (northern Chukchi Sea shelfbreak), shipboard suspended underwater sound sources were deployed to transmit acoustic signals to hydrophone arrays moored on the deep basin as well as the shallow shelf. In this paper, numerical simulations utilizing the Parabolic Equation method are conducted to provide physical insights into the variability of signals propagating over the shelfbreak and slope and recorded on a vertical array on the Chukchi Sea shelf. The numerical models simulate sound propagating over the slope via the Pacific Halocline duct, which is a water-borne vertical sound duct formed between the layers of Pacific summer water and Atlantic water. The source to receiver distance is about 130 km, and realistic variability is introduced in the numerical models. The previous studies reported in the literature have concluded that the shelfbreak circulation, specifically upwelling, and the submesoscale eddies spun off the shelfbreak jet are the two major causes of the temporal and spatial variability of the Pacific Summer Water Layer over the slope in the region. The numerical simulation study also emphasize on the scattering and attenuation effects caused by ice cover and roughness. A preliminary data-model comparison is made and discussed in this paper. [Work supported by the Office of Naval Research.]

3:20

4pUW7. Potential impacts of climate change on acoustic propagation in the Arctic. Timothy F. Duda, Lee E. Freitag (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), Lori R. Adornato (SRI Int., St. Petersburg, FL), and Robert H. Byrne (College of Marine Sci., Univ. of South Florida, St. Petersburg, FL)

Some forecasts show surface pH in the Arctic dropping from 8.1 to 7.6 over the next 100 years. This substantial decrease may cause changes in acoustic transmission at frequencies where the pH-dependent borate absorption plays a role, below about 5 kHz. In many cases, upward refraction of sound in the near-isothermal Arctic waters causes ice or surface scattering effects to dominate transmission. However, recent observations in the Canada Basin show that 700 and 900-Hz sound can be fully ducted beneath the Pacific Summer Water, with no ice interaction, and be detectable over distances of a few hundred kilometers. In this situation, the received signal level is controlled largely by cylindrical spreading and absorption. Here, for a wide band of frequencies, the effects of probable pH reductions and reduced absorption are investigated using a few models of pH depth profiles. There is potential for increased signal levels of 5 dB or more for 200-km propagation if the duct waters have significantly reduced pH.

3:35

4pUW8. Implementation of moving magnet actuation in very low frequency underwater acoustic transduction. Brenton Wallin (Sensors & SONAR Systems Dept., Naval Undersea Warfare Ctr., 30 Summit Ave., Narragansett, RI 02882, brenton.wallin@navy.mil), Steven Crocker, and Jeffrey Szelag (Sensors & SONAR Systems Dept., Naval Undersea Warfare Ctr., Newport, RI)

The Naval Undersea Warfare Center's Underwater Sound Reference Division (USRD) is designing a very low frequency sound source to calibrate line arrays in the 1–100Hz band at their Leesburg Facility. For calibrations below 20 Hz, the low frequency cutoff of their J15-3 standard projector, the USRD currently uses a passive method where the sound source is the ambient noise in the spring. A limitation of this method is a low signal-to-noise ratio. By designing, building, and implementing the aforementioned sound source, the USRD looks to increase the signal-to-noise ratio of their

recorded data, and consequently increase the reliability of measurements at very low frequencies. This presentation explores the idea of implementing moving magnet technology in a very low frequency transducer. Although moving coil projectors have been used since 1914 as low frequency sound sources, their acoustic output at very low frequencies is limited. The potential benefits of a moving magnet projector over moving coil projectors are discussed. Acoustic performance estimates for a moving magnet projector are calculated based on specifications of an off-the-shelf moving magnet actuator. Lastly, the presentation will cover current efforts being made to design an underwater housing for the projector.

3:50

4pUW9. Reducing numerical dispersion and reflections in finite element and finite difference simulation of acoustic wave propagation and scattering. Murthy Guddati (NC State Univ., 2501 Stinson Dr., NCSU-Civil Eng., Raleigh, NC 27695-7908, mnguddat@ncsu.edu) and Senganal Thirunavukkarasu (Enthought, Inc., Austin, TX)

Discretization methods such as the finite element method (FEM) and the finite difference method (FDM) suffer from two types of accuracy problems: spurious wave dispersion for long-range propagation and artificial reflections from discretization of perfectly matched layers that are often used to represent the unbounded exterior. In this talk, we focus on lower-order FEM and compact FDM stencils, and show that these errors can be significantly reduced using simple measures: using modified integration rules for FEM, and modified compact stencils for FDM. It turns out that these needed corrections for reducing dispersion and reflections are in opposite directions, indicating that reducing one error results in increasing the other, which is not desirable. In this talk, we introduce a special technique that leads to reduction of both dispersion and reflection errors, and illustrate its effectiveness through theoretical analysis and numerical experiments.

THURSDAY EVENING, 26 MAY 2016 — 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Thursday are as follows:

Committee	**Start Time**
Noise	Salon D
Speech Communication	Salon G
Underwater Acoustics	Salon J

Session 5aAA

Architectural Acoustics: A Variety of Interesting Research and Observations in Architectural Acoustics

David S. Woolworth, Chair
Oxford Acoustics, 356 CR 102, Oxford, MS 38655

Chair's Introduction—8:00

Contributed Papers

8:05

5aAA1. A potential new and better method for measuring transmission loss in the field. Paul D. Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

In a recent study, transmission loss (TL) measurements were made from outdoors-to-indoors and indoors-to-outdoors of a house. These results agree with one another within 0.6 dB. The agreement achieved in this recent study is believed to be because the indoor measurements for the indoor-to-outdoor TL were made at the party-wall surface of the reverberant space. This current paper demonstrates what amounts to a form of pressure doubling at the surfaces of the room containing the reverberant field. It is this higher level that must be used in the TL calculation from indoors-to-outdoors; not the reverberant field measured interior to the room. The actual increase in level for this reverberant-field pressure enhancement appears to be close to + 2.7 dB, which is consistent with measurements of free-field pressure doubling on a hard surface, which are theoretically + 6 dB, but typically measured to be + 5 to + 5.5 dB. This factor of + 2.7 dB for reverberant-field pressure doubling should be applicable to all measurements that use a reverberant space such as laboratory facilities that measure transmission loss from a reverberant room to a more absorptive space.

8:20

5aAA2. Comparison of interior noise levels produced by rain impinging on several commercial roof constructions. Logan D. Pippitt, Michelle L. Huey, and Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, ldpippitt@gmail.com)

Noise caused by rain on commercial roofs is an ever-occurring problem. Due to climate changes, rainfall seems to have become more sporadic and intense, making the effects of rain noise on the interior environment an acoustical issue of intensifying importance. Noise produced within architectural spaces by rain on roofs is difficult to quantify due to varying rain intensity, water droplet size, droplet velocity, roof construction, and interior acoustical characteristics. However, it is possible to compare interior rain noise levels when rain conditions and the interior space conditions are constant. This paper compares (1) rain noise levels for several commercial roof constructions with and without suspended ceilings beneath the roof, (2) rain noise levels with roofs using typical rigid foam insulation versus mineral wool insulation. Noise level measurements and rain generation are in general conformance with ISO 140-18 "Laboratory measurement of sound generated by rainfall on building elements." The rain noise levels are presented along with similar rain noise measurements made in the same test facility that were the subject of a 2007 ASA paper. This research was conducted through the School of Architecture, Design and Planning at the University of Kansas.

8:35

5aAA3. Design, construction, and evaluation of a binaural dummy head. Maryam Landi, Vahid Naderyan (Dept. of Phys. and Astronomy & National Ctr. for Physical Acoust., Univ. of MS, NCPA, 145 Hill Dr., University, MS 38677, mlandi@go.olemiss.edu), and David S. Woolworth (Roland, Woolworth & Assoc., Oxford, MS, Oxford, MS)

Binaural-dummy-heads are often used as standard measurement devices where modeling of the human binaural hearing system is desired. The binaural-dummy-head imitates a human head (and torso) which is used in binaural recording as well as research areas such as hearing aids, sound localization, noise measurements, etc. Commercially available binaural heads are not economically efficient for some purposes. This paper outlines a less expensive binaural dummy head built using ANSI/ASA S3.36-2012 standard as a reference as part of an independent coursework. A hard plastic mannequin was used as head and torso, and the two ears were real human ear replicas casted out of water-based alginate gel. The complex Head-Related Transfer Functions (HRTF) of our dummy head were measured in an anechoic chamber to evaluate its spectral and directional properties and were compared to the same properties of the standard commercial dummy head.

8:50

5aAA4. ISO 717-1 introduction in Russia. Ilya E. Tsukernikov, Igor Shubin (Acoust. Lab., Res. Inst. of Bldg. Phys., Odoevskogo proezd, h.7, korp. 2, fl. 179, 21 Lokomotovny pr., Moscow 117574, Russian Federation, 3342488@mail.ru), Tatiana Nevenchannaya (Moscow State Univ. of Printing Arts, Moscow, Russian Federation), and Natalia Schurova (Acoust. Lab., Res. Inst. of Bldg. Phys., Moscow, Russian Federation)

Two single-number quantities are used now in Russia: weighted sound reduction index for rating air noise insulation by internal protecting designs of buildings and the quantity, characterizing sound insulation of external transparent protecting designs from noise, created by municipal transportation streams. For other sound insulation characteristics determined by International Standards ISO 10140-2, ISO 140-4, and ISO 140-5, the corresponding single-number quantities are not established in the Russian standard documents. Besides methods for determination of single-number quantities to be used differ by their various physical senses too. It causes expediency of introduction of International Standard ISO 717-1 in which the corresponding single-number quantities are entered for all spectral characteristics, put into practice of air noise insulation by building protecting designs, and universal methods of their determination are stated. In this paper, features of introduction of International Standard ISO 717-1 in Russia are considered. Comparison of the A-weighted noise spectra for various categories of the railway transportation, to be maintained on the Russian railways, to the sound level spectra, applied in the International Standard to calculate the spectral adaptation terms. By the concrete examples, the divergences in values of spectral adaptation terms received are shown and the corresponding recommendations are offered.

9:05

5aAA5. Characterization of ensemble rehearsal experiences at Brigham Young University. Kieren H. Smith, Tracianne B. Neilsen, Michael H. Denison, and Jeremy Grimshaw (Brigham Young Univ., Provo, UT 84602, kierenhs@gmail.com)

Musicians' ears are barraged with large quantities of sound almost constantly, a reality that is either augmented or diminished by room environments in which musicians practice and perform. In order to make recommendations for future renovations, the acoustics of ensemble rehearsal spaces within the Brigham Young University School of Music were measured. To quantify sound exposure during rehearsals, noise dosages and sound levels experienced by the musicians were measured in various positions in each of two major practice spaces. Measurements were taken during several two-hour rehearsals for major orchestras and band ensembles at BYU. Using data collected from noise dosimeters, spatial maps indicate the noise dosage throughout the rooms. Maximum and average sound levels experienced during the space of a rehearsal also offer insights into the sound environment. This data indicate which areas of the ensemble experience the greatest noise exposure. Reverberation time measurements taken within the rooms further illuminate potential acoustic deficiencies within the room. This preliminary noise environment and dosage highlights the need for future facility renovations and offers suggestions for short term acoustical treatments. EASE acoustic simulation software was used to determine the effectiveness of suggested renovations.

9:20

5aAA6. Reverberation theory obscures real physics in concert halls. James B. Lee (None, 6016 S. E. Mitchell, Portland, OR 97206, cadwal@macforcego.com)

Wallace Sabine posited sound in a room to be a uniform field, in equilibrium, varying but slowly with respect to time required to traverse the space: the "reverberant" field. It easy to demonstrate that such is not so, especially in rooms like occupied concert halls. But then, Sabine designed the Boston Symphony Hall, a paradigm of acoustic excellence. If that hall be regarded as a physics experiment, never has it been replicated by Sabine's followers, even with extensive emendations to his concept of reverberation. The real physics of concert halls involves non-equilibrium manifestation of physical acoustics with respect to bounding surfaces, particularly proximity effects on stage and resonant scattering about the audience.

9:35

5aAA7. Evaluating hospital quiet time from engineering, medical, and nursing perspectives. Jonathan R. Weber, Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, jonryanweber@gmail.com), Ashley Darcy Mahoney (Nell Hodgson Woodruff School of Nursing, Emory Univ., Atlanta, GA), Myra Rolfes, Heather Cooper (Neonatal Intensive Care Unit, Children's Healthcare of Atlanta, Atlanta, GA), and Brooke Cherven (Nursing Res. and Evidence Based Practice, Children's Healthcare of Atlanta, Atlanta, GA)

A healthy hospital soundscape is crucial to promote healing for patients and a healthy workplace for staff. Unfortunately, the occupant-generated sounds, building systems, and medical equipment required to care for patients can create high noise levels. Rising concern has led to an increase in hospital noise studies exploring noise reduction strategies. One major issue is the gap existing between the acoustical and medical contributions necessary to solving the problem. This talk will document how our team is focusing on bridging that gap through the interdisciplinary collaboration of individuals from architectural engineering, medicine, nursing, psychology, and statistics. The project includes an 18-month longitudinal study aiming to improve Neonatal Intensive Care Unit (NICU) soundscapes through the implementation of a Quiet Time (QT) evidence-based practice change. Detailed acoustic measurements and staff surveys were collected to document the objective and subjective effects of QT. The acoustical impact of QT on the soundscape and its occupants is currently being explored through the engineering and medical perspectives. In ongoing phases, infant physiological data are being analyzed to understand the infants' response to the altered soundscape resulting from QT. The collaborative efforts required to plan, execute, and evaluate this type of interdisciplinary study will be discussed.

9:50

5aAA8. Death by alarm: An error model of hospital alarms. Ilene Busch-Vishniac (School of Nursing, Johns Hopkins Univ., 200 Westway, Baltimore, MD 21212, buschvi@gmail.com)

On any given day in the United States, there are about 480,000 patients in hospital for reasons other than psychiatric care or rehabilitation, each generating, on average, about 135 clinical alarms per day. Studies have shown that over 90% of these alarms result in no action being taken. Alarm errors, either alarms that sound and receive no response or alarms that fail to sound when they should, number roughly 8 million per day yet data on adverse alarm impacts indicate about 200 alarm-related deaths per year and a total of a little more than 500 adverse impacts per year. A compelling conclusion from this data is that clinical alarms in hospital are very inefficient and ineffective tools for monitoring medical emergencies. Much attention has been dedicated to alarms recently, with the general goal of improving response to alarms in order to ensure no medical emergency is missed. While this work is of immediate use and is vitally important to the operation of the modern hospital, it focuses on minor changes to the existing systems rather than on trying to design the optimum system for the future. It is future alarm systems that we consider here, with an aim of designing a more effective and efficient system for use in hospitals in roughly 20 years.

10:05–10:20 Break

10:20

5aAA9. A method to evaluate nonideal effects of anechoic chambers on multiple-angle measurements. Michael H. Denison, K. J. Bodon, and Timothy W. Leishman (Phys., Brigham Young Univ., 485 S State St., Apt. 306, Provo, UT 84606, michael.denison23@gmail.com)

Anechoic chambers are typically qualified by comparing sound pressures at several radial distances from a sound source and verifying that they follow the spherical spreading law within specified tolerances. While this technique is useful, it may not sufficiently characterize free-field variations at fixed radial distances and numerous angular positions, as are commonly used for directivity, sound power, and other important acoustical measurements. This paper discusses a technique to detect angular field deviations in anechoic chambers. It incorporates a loudspeaker in an altazimuth mount, an adjustable-radius boom arm, and a precision microphone. The boom arm and microphone remain in line with the loudspeaker driver axis at a fixed radius while the system rotates to specified polar or azimuthal angle increments. In an ideal free-field environment, the frequency response function from the loudspeaker input to the microphone output should remain consistent—regardless of the system orientation. However, in typical anechoic chambers, they vary. Standard deviation calculations over many angles reveal frequency-dependent departures from the ideal, especially for narrow-band data. The results show the impact of these discrepancies for multiple-angle measurements and how they change with radial distance from the source.

10:35

5aAA10. Unexpected challenges of luxury residential building acoustics. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

Building acoustics involving sound enhancements, noise abatement, and vibration control for luxury residential properties can present more than just technical, performance, and mitigation issues. The normal noise control issues dealing with mechanical, electrical, and plumbing (MEP) systems are common in commercial and multi-family residential projects. However, ultra-expensive, highly customized, single-family residences are a very different situation, both from the expectation of quality and the much larger and more varied types of noise producing systems. Such custom homes of entertainment and movie stars, corporate moguls, and sports personalities include features and systems that would never occur in the typical single-family home. Large in-home theaters with fully outfitted sound systems, indoor swimming pools, large entertainment centers, dance and reception halls, etc., present very challenging acoustical and noise control challenges. Many such estates also have commercial or industrial type mechanical systems, multi-fueled emergency power generators, and very large water storage and pumping systems that

generate significant noise control issues both within the owner's property and for adjacent neighboring properties. This paper summarizes a few such challenging projects and reviews the often unusual situations that occur during concept and project design, specification and bidding phases, construction, redesign, change-orders, and essential inspections.

10:50

5aAA11. Acoustical evaluation of residential laneways and laneway housing. Rosa Lin and Maureen Connelly (School of Construction and the Environment, Br. Columbia Inst. of Technol., 3700 Willingdon Ave., NE03, Burnaby, BC V5G 3H2, Canada, rlin35@bcit.ca)

Laneway housing (LWH), a free-standing, small wood-frame house (1 to 1.5 stories high and less than 900sf in floor area), is an increasingly popular product of Vancouver B.C.'s high-density urban growth policy. LWH is subject to multiple noise concerns due to its small architectural form and siting in laneways typically designed for garage access and utilities. (Laneways are approximately 16 ft wide and 12 to 18 ft high.) Case study investigations examined residential laneways as potential urban canyons and the acoustical performance of LWH envelope. Methods employed include ASTM field measurements, software models (Odeon and AFMG Soundflow), and the National Research Council of Canada traffic noise model (comparable to the FHWA model). Metrics evaluated include rate of attenuation over propagation distance through a laneway, façade transmission loss, and room absorption. Results show that certain laneways function as urban canyons, measuring nearly 20 dBA higher in SPL at the LWH façade than in laneways without urban canyon characteristics. Indoor SPL due to traffic noise transmitting through the building envelope were above 45 dBA. At least half of the case studies investigated did not meet noise criteria for residential health. These investigations emphasize the need for acoustical specifications in design and construction guidelines for LWH.

11:05

5aAA12. A novel method for perceptual assessment of small room acoustics using rapid sensory analysis. Neofytos Kaplanis (Bang & Olufsen, Peters Bang vej 15, Struer, Denmark, neo@bang-olufsen.dk), Søren Bech (Electron. Systems, Aalborg Univ., Struer, Denmark), Tapio Lokki (Dept. of Media Technol., Aalto Univ., Helsinki, Finland), Toon van Waterschoot (Elec. Eng., ESAT-STADIUS/ETC, KU Leuven, Leuven, Belgium), and Søren H. Jensen (Electron. Systems, Aalborg Univ., Aalborg, Denmark)

Identifying and perceptually characterizing the physical properties of rooms is a fundamental step in understanding the acoustical qualities of a space. Over the last century, numerous studies have investigated the perceptual qualities in performance spaces, such as opera houses and concert halls. In smaller spaces, such as domestic environments, the research focus has been primarily steered toward sound reproduction within a room, rather than the transmission medium, the room. In this study, a new methodology is used to perceptually assess and characterize a range of acoustical properties within small rooms and car cabins. In-situ measurements were performed to obtain a range of possible acoustical settings, by varying physically the spaces under investigation. The measured responses were spatially analyzed and synthesized to reproduce the observed fields in the laboratory. Expert listeners were presented with auralized sound over a loudspeaker array and followed a rapid sensory analysis protocol. The elicited attributes and ratings are analyzed and possible links to the acoustical properties of these spaces are discussed. [*This study is a part of Marie Curie Network on Dereverberation and Reverberation of Audio, Music, and Speech. EU-FP7 under agreement ITN-GA-2012-316969.]*

11:20

5aAA13. Investigation of coprime scattering arrays as sparse sound diffusers. Kevin A. Melsert, Dane Bush, and Ning Xiang (Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, melsek@rpi.edu)

Number-theoretic coprime sparse samplers have recently sparked research activities in the fields of signal processing and acoustics. The concept of coprime sparse sensing has been successfully implemented and experimentally tested in form of acoustic sparse microphone arrays, as shown in the publication [D. Bush and N. Xiang, J. Acoust. Soc. Am. 138, 447-456 (2015)]. The current study investigates how coprime theory can be applied to scattering incident sound. It also investigates how coprime array concepts can be applied to volumetric diffusers using solid cylinders as sparse array elements. These are arranged in pairs of subarrays whose number of cylinders are two mutually prime (coprime) integers in a sparse arrangement. Coprime scatter arrays are tested experimentally as sparse sound diffusers using one-fifth scale modeling and an acoustic goniometer to investigate coprime sparse diffuser effectiveness in real world applications.

11:35

5aAA14. Empirical measure of absorption and scattering properties of living wall plants and systems and predictive modeling of room acoustic benefits. Maureen R. Connelly, Daver Bolbolan, Masha Akbarnaejad, and Sepideh Daneshpanah (Construction and Environment, BCIT, 3700 Willingdon Ave., Bldg. NE03 Office 107, Burnaby, Vancouver, BC V5G3H2, Canada, maureen_connelly@bcit.ca)

This series of research projects investigate the acoustical characteristics of interior living walls and predicts how they can be used to positively benefit room acoustics. Scaled and full scale evaluations were executed in a reverberation chamber to validate test methods for absorption and scattering coefficients of soil/substrate and plant species (characterized by height, stem diameter, mass, leaf geometries and dimensions, and leaf area index (LAI)). Wall systems were evaluated over a gradient of plant coverage with monoculture and community planting. Evaluation indicates that evenly distributed pumice can act as the baseline on the scattering turn-table in a method to evaluate scattering coefficients of plant-specific foliage. Findings indicate that percentage of plant coverage is related to absorption coefficients (0.16–1.1) as averaged across all evaluated species. Only at low plant coverage do specific plant characteristics affect the absorption coefficients. The percentage of plant coverage is related to scattering coefficients (0.05–0.51) at 500 Hz and higher. LAI x Mass predicts absorption and scattering coefficients at mid-frequency (200–2500 Hz). Comparison of prediction and field studies identify that use of scattering coefficients improves the prediction of the beneficial use of living walls in room acoustics.

11:50

5aAA15. A hybrid method combining the edge source integral equation and the boundary element method for scattering problems. Sara R. Martín Román, U. Peter Svensson (Acoust. Res. Ctr., Dept. of Electronics and TeleCommun., Norwegian Univ. of Sci. and Technol., O.S. Bragstads plass 2a, Trondheim 7034, Norway, sara.martin@ntnu.no), Jan Šlechta (Univ. Ctr. for Energy Efficient Buildings, Czech Tech. Univ. in Prague, Bustehrad, Czech Republic), and Julius O. Smith (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA)

A hybrid method for acoustic scattering problems is studied in this paper. The boundary element method is combined with a recently developed edge diffraction based method [J. Acoust. Soc. Am. **133**, pp. 3681–3691, 2013]. Although the edge diffraction method has been shown to provide accurate results for convex, rigid objects at a very attractive computational cost, it has some numerical challenges for certain radiation directions. The hybrid method suggested here has a similar structure as the boundary element method (BEM): in a first step, the sound pressure is calculated on the surface of the scattering object, and in a second step, the scattered sound is obtained at any external receiver point. In this method, the edge diffraction based method is used for the first step, and then, the calculation of the scattered sound is performed à la BEM by means of the Kirkhoff–Helmholtz Integral equation. Several benchmark cases are studied, and the results are compared with different reference methods.

Session 5aMU

Musical Acoustics: General Topics in Musical Acoustics

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Contributed Papers

8:00

5aMU1. Real-time three-dimensional tongue motion during clarinet performance. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu), Sherman Charles (Linguist, Indiana Univ., Bloomington, IN), and Benjamin Lulich (The Cleveland Orchestra, Cleveland, OH)

Clarinetists teach and feel that they manipulate their tongue shape and position in order to properly “voice” music with ideal intonation and timbre, as well as for special effects such as portamento (pitch bending) and glissando. Two basic postures are typically described, which are frequently called the “ee” (or “er”) and “ah” positions, or “voicings,” due to their presumed similarity to tongue shapes during production of the speech sounds [i] (or [r]) and [a]. Although some two-dimensional imaging studies of tongue shape during clarinet performance have been reported, three-dimensional (3D) data have not been previously available. This study presents 3D tongue motions during performance by one professional clarinetist. Tongue images were acquired from a 3D/4D ultrasound system with synchronized audio recordings, and the images were aligned with a digitized impression of the performer’s palate. Analyses of the data are ongoing, but initial results indicate that the “ee” and “ah” postures are based on the tongue shape but not on the tongue position. Further aspects of the tongue shape and position during clarinet “voicing” will also be discussed.

8:15

5aMU2. Resonant mode characteristics of a cajón drum and its effect on sound directivity. Michael H. Denison, K. J. Bodon, and Kent L. Gee (Phys., Brigham Young Univ., 485 S State St., Apt. 306, Provo, UT 84606, michael.denison23@gmail.com)

The cajón is a hand percussion instrument originally from Peru. Spanish for “large box” or “crate,” the cajón is typically a box enclosure with a thin wooden plate at the front that serves as the playing surface. Many resonance modes are excited when a player strikes the front plate, as recently studied by Pehmoeller and Ludwidgsen [J. Acoust. Soc. Am. **138**, 1935 (2015)]. In this study, scanning laser Doppler vibrometry is coupled with high-resolution far-field directivity measurements to examine source vibration and the resultant sound field. Ties between the plate modal vibration and far-field sound radiation spatial patterns are shown. In particular, the non-uniform front plate normal velocities at higher-order resonances result in distinct nulls in the far-field directivity. Avoidance of these nulls yields recommended microphone locations for cajón sound amplification.

8:30

5aMU3. Investigating the effect of body geometry on the acoustics of electric guitars. Mark Rau and Gary Scavone (Music, CAML - McGill Univ., 2325 Maisonneuve West, Apt. 2, Montréal, QC H3H1L6, Canada, mark.rau@mail.mcgill.ca)

Three electric guitars of different body geometries are investigated in an attempt to characterize their tonal differences. One guitar has a solid body, one is fully hollow, and one is semi-hollow which represents a midway point between the first two. The transfer functions of the electromagnetic pickups are determined by inducing a known signal through the pickup and measuring the output. Input admittance measurements are taken at the bridge and nut of each guitar to show the vibrational modes of the bodies. Wire break measurements are taken for different notes on each guitar, which are analyzed in conjunction with the admittance measurements. The results demonstrate significant admittance peaks in the hollow and semi-hollow guitars that are not present in solid body electric guitars. As well, “dead” notes with shorter decay times are found to be correlated with these admittance peaks, thus indicating that the bodies of hollow and semi-hollow electric guitars play a role in their tonal characteristics.

8:45

5aMU4. Instrument identification and blending in vinyl records during the transition period from jazz to rock music. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Arnold Schering and others have partitioned music genres into traditions where the identification of individual instruments (Spaltklang) is the goal and others where instruments are blended to a homogeneous sound (Verschmelzung). Baroque music is often cited to emphasize the identifiability of individual instruments, whereas the fusion of orchestral instruments became one of the major goals in the romantic period. Historical vinyl record releases were used to investigate the extent to which the theory of identification and blending can be applied to 20th century popular music based on the fundamental principles of Auditory Scene Analysis. The identifiability of instruments was one of the major concerns for jazz combos, for example, in the Miles Davis sextet of the late fifties. Blending became more important in many types of rock music to create a huge sound, for example, in Phil Spector’s Wall of Sound of the 1960s. Artificial reverberation, dynamic range compression, and other effects were now used to create loud and fused sound fields. Based on a psychophysical loudness model, it will be discussed how both approaches lend themselves to create loud sounding mixes to maximize the achieved perceived loudness for a given medium-dependent maximum signal level.

9:00

5aMU5. Vibration analysis of acoustic guitar string employing high-speed video cameras. Bozena Kostek (AudioAcoust. Lab., Gdansk Univ. of Technol., Natowicza, 11/12, Gdansk, Pomorskie 81-233, Poland, bokostek@audioacoustics.org), Piotr Szczuko, Jozef Kotus, Maciej Szczodrak, and Andrzej Czyzewski (Multimedia Systems Dept., Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland)

A method of analysis and visualization of displacements of an acoustic guitar string is presented. Vibrations of the strings are recorded using high-speed cameras. The optical system used for the recording is applied in order to make it possible to observe the vibrations along the string. Images recorded with high-speed cameras are analyzed using digital signal processing algorithms in order to track the shape of deflections and displacement of strings, with a high spatial resolution, and to convert the acquired video data into an acoustic signal. The acoustic signal derived from the visual analysis is then compared with a reference signal which was recorded simultaneously using a measurement microphone. The research experiments are aimed principally at studying the phenomena related to energy transfer of vibrating strings to the body of the instrument. [This research study was supported by the grant, funded by the Polish National Science Centre, decision number DEC-2012/05/B/ST7/02151.]

9:15

5aMU6. A physical model of a highly nonlinear string and its use in the music composition *Quartet for Strings*. Edgar Berdahl, Stephen D. Beck, and Andrew Pfalz (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

A real-time virtual string model is created that supports two modes of haptic force-feedback interaction. Using one haptic controller, the string can be plucked, and using a second haptic controller, the virtual string's pitch can be continuously varied by applying pressure to the string. Via both controllers, the vibrations of the virtual string can be felt while it is played. These functionalities are achieved by modeling the string as a finite sequence of interleaved masses and stiffening springs. For artistic purposes, a nonlinear spring force characteristic function is selected that limits the effective stiffness of each spring to a range between k and $(k+k_s)$. This function is $F(x) = kx + (k_s x^3)/(\beta^2 + x> + x^2)$, where x is the displacement of the nonlinear spring and β approximately adjusts the displacement at which the stiffness becomes significantly nonlinear. For small displacements, the string's undamped fundamental frequency is tuned by k, and for sufficiently large displacements, the string's undamped fundamental frequency is approximately tuned by $(k+k_s)$. The composition *Quartet for Strings* features the model tuned to four different scales (treble 1, treble 2, alto, and bass), each of which is played by a human performer according to the score of *Quartet for Strings*. For this composition, $(k+k_s)$ is set so that when the string is vibrated at sufficiently large displacements, the pitch increases by approximately an octave but not precisely an octave, in order to provide for a more complex pitch space.

9:30

5aMU7. Predicting the acoustical properties of 3d printed resonators using a matrix of impulse responses and mode interpolation. Romain Michon (Dept. of Music, Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., Stanford, CA 94305-8180, rmichon@ccrma.stanford.edu) and John Granzow (School of Music, Theatre & Dance, Univ. of Michigan, Ann Arbor, MI)

Accurately predicting acoustical properties of 3D printed models is of interest to instrument designers who explore novel geometries. We introduce a technique to carry out these estimates using a database of impulse responses and mode interpolation. 3D models are organized as a function of their physical characteristics and placed into a multidimensional space/matrix. The models at the boundaries of this space define the limits of our prediction algorithm and they are produced using 3D printing. Impulse responses of these models are measured, and modal information is extracted from each object. Mode parameters are interpolated within the matrix to predict the frequency response of unprinted models that fall within the geometrical space of the test matrix. A physical model using modal synthesis also allows us to listen to the resulting resonator.

9:45

5aMU8. Physical modeling sound synthesis using embedded computers: More masses for the masses. Edgar Berdahl and Matthew Blessing (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, edgarberdahl@lsu.edu)

Physical modeling for sound synthesis is a technique in which musical acoustic equations are simulated by computer to synthesize sound. In prior decades, either offline simulation or powerful desktop or laptop computers were required in order to synthesize high-quality sound. However, increasingly small and relatively low power embedded computers are presently becoming available that can natively perform real-time simulations using floating-point computations. For example, the Raspberry Pi 2 is an embedded computer, which incorporates a quad-core 1GHz embedded microprocessor, and currently costs only US$35. This implies that physical modeling sound synthesis may become accessible to a wide range of people for many diverse applications. Furthermore, larger and larger numbers of virtual masses will be computable in real time. A poster with embedded Raspberry Pi 2 and amplified speaker is presented that uses Synth-A-Modeler to simulate a wide variety of physical models in real time.

Session 5aNS

Noise: Topics in Noise Control

Blaine M. Harker, Chair
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Contributed Papers

8:30

5aNS1. Aircraft carrier noise measurements of a high-performance fighter jet. Alan T. Wall, Richard L. McKinley, Michael R. Sedillo, Billy J. Swayne (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Michael J. Smith, Allan C. Aubert (NAVAIR, Pax River Naval Base, MD), Robert E. Nantz, and Gregory J. Imhof (Joint Strike Fighter Integrated Test Force, Pax River Naval Base, MD)

The on-deck measurement of F-35C noise levels occurred during the DT-II sea trials aboard the USS Dwight D. Eisenhower in October 2015. The existence of aircraft carrier flight deck fighter noise data is extremely rare, with this data set being only the second of its kind in terms of its scope. Custom acoustic recording instrumentation was designed to obtain quality broadband noise measurements of high-amplitude signals, protected from extraneous noise due to high wind speeds, and shielded from intense electromagnetic interference from the multiple on-board radar systems. The data collected allow for the estimation of noise exposures at all pertinent flight deck locations where crewmembers are positioned. [Work supported by USAFRL through ORISE and F-35 JPO.]

8:45

5aNS2. Review study of main sources of noise generation at wheel-rail interaction. Qinan Li and Mohammad Mehdi Alemi (Mech. Eng., Virginia Tech, 1100 Houndschase Ln., Unit G, Blacksburg, VA 24060, liq001@vt.edu)

Railways are very economical and efficient long-distance transportation systems in many high-density populated countries. However, in terms of environmental safety, different countries announced an increasing traffic noise near residential area. The relative importance of wheel-rail noise to overall train noise plays a crucial role in developing railway transportation. Generally, there are three types of noise at wheel-rail interaction: rolling noise, impact noise, and squeal noise. Rolling noise refers to vertical vibration excitation on straight track due to the undulations of the wheel and rail surfaces. TWINS as the most advanced model up to date can provide total noise prediction within 2 dB compared to full-scale experiments. The impact noise happens when wheel is running on the rail surface discontinuities. Since linear noise generation assumption is employed at contact patch and non-linear contact element is ignored, impact noise due to large roughness are rarely developed and they need to be extended to the reasonable higher frequencies (5 kHz). The squeal noise is usually due to lateral excitation mechanism and always occurs on sharp radius curves. Due to the complexity of wheel-rail noise generation mechanisms, the almost all existing prediction models either have not been validated or they are bounded to predict noise under limited conditions and certain frequency ranges. In this paper, the differences between three main sources of noise generation at wheel-rail interaction will be reviewed.

9:00

5aNS3. A study on minimizing the noise from a trailing edge by the use of optimized serrations. Matt Brezina and Joana Rocha (Mech. and Aerosp. Eng., Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, matt.brezina@carleton.ca)

An optimization study is presented in which the noise produced by flow over the trailing edge of a flat plate is minimized via serrations. The theoretical optimum configuration is found by changing the geometry of the serrations with a non-linear gradient based algorithm. The theoretical noise produced by the trailing edge is determined using the semi-empirical model developed by Howe. As expected, the configuration of the trailing edge serrations that produced the least noise over all frequencies (from 20 Hz to 20 kHz) was found to be those with the smallest width (approaching zero) and largest root-to-tip distance (up to the assigned upper-bound). The optimum serration geometry for individual frequencies was found to be highly dependent on the upper bound given to the height. In addition to the optimization study, an analysis is presented of a method for modifying the noise model to capture the high-frequency noise increase (relative to a straight trailing edge) that is regularly observed in experiments. Results are presented for model validation by a comparison to the sound pressure level frequency spectrum produced experimentally.

9:15

5aNS4. Modeling systems of acoustic resonators for application in passive noise control. Matthew F. Calton and Scott D. Sommerfeldt (Brigham Young Univ., 560 E 400 N APT 4, Provo, UT 84606, mattcalton@gmail.com)

Acoustic resonators, such as Helmholtz and quarter wave resonators, are commonly used as narrowband attenuators in small enclosures. The widespread use of resonators has led to many analytical expressions and approximations for their response, especially for the ideal geometry case. While these resonators are limited in bandwidth, systems of resonators can be designed to obtain responses not otherwise attainable by a single element. Additionally, practical applications of these systems often require non-ideal geometries to be implemented in the design process. This research aims to incorporate equivalent circuit techniques to more accurately characterize the input impedance of a system of resonators. Using the calculated input impedance of the system, an optimal coupling location is determined using a model of the coupled source and enclosure. These calculations are compared to experimental results for validation.

9:30

5aNS5. Active control of noise radiated from an x-ray tube. Yin Cao, Kelli Succo, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N203 ESC, BYU, Provo, UT 84602, kelli.fredrickson7@gmail.com), and Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

The application of implementing an active noise control system to globally attenuate noise radiated from a medical x-ray tube has been investigated. The noise radiated from the x-ray tube is characterized by the presence of numerous tonal peaks distributed over a broad frequency bandwidth. Furthermore, there is sufficient variability in the radiated field that

coherence in the acoustic field was also found to be a challenge. It was determined that a properly placed structural sensor could be used as a reference signal to achieve good coherence between the reference signal and error microphone. In order to achieve global control, speakers were placed in close proximity to areas that had been identified as source regions through the use of SLDV and acoustic intensity measurements. The frequency band below 1500 Hz was targeted, and it was found that effective attenuation could be achieved at over ten of the most prominent frequencies, resulting in attenuations in that bandwidth on the order of 7 dB. Some of the challenges encountered and results obtained will be discussed and presented.

9:45

5aNS6. Active control of a finite length line source by a novel secondary source. Qi Hu and Shiu-keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., ZS801, Block Z, Hung Hom Na, Hong Kong, qi.bs.hu@connect.polyu.hk)

Active control of noise in free space tends to be challenging, especially for the overall attenuation. This paper focuses on the control of a finite line noise source using active approach. Previous study by the author reveals that the introduction of a directional source, an axially oscillating circular piston, as the secondary control source will improve the overall performance of the active noise control system. A novel directional source is theoretically studied in this work, which consists of a core piston and an outer concentric annulus. The two components are axially oscillating with different phases and amplitudes, and these parameters are optimized to maximize the source directivity. This structure can be further extended to add more outer annuluses, which also could be analytically analyzed of the optimal parameters according to the similarity of the construction components. This exquisitely novel control source is adjusted to radiate a sound field analogous to the primary line source with multi-lobe property, and its physical size is restricted within a practical dimension.

10:00

5aNS7. Development of a direct aerodynamic drag measurement system for acoustic liners. Christopher Jasinski (Univ. of Notre Dame, 54162 Ironwood Rd., South Bend, IN 46635, chrismjasinski@gmail.com)

The objective of this paper is to establish the validity of a measurement apparatus for determining aerodynamic drag caused by acoustic liners. Conventional acoustic liners for ducted turbofan engine nacelles have been proven to reliably reduce engine noise in commercial aircraft for a narrow frequency range. Traditional acoustic liners consist of a porous facesheet, honeycomb core, and solid backing. As technology has developed, new liner designs, including variable-depth cores, show promise of reducing noise in a broad frequency range. Additionally, the next generation of aircraft design may allow for additional aircraft surface area to be covered by acoustic liners, further reducing aircraft noise perceived on the ground. Before these advanced acoustic technologies can make their way onto commercial aircraft fleets, the aerodynamic drag caused by the liners must be understood. Using the Mach 0.6 Wind Tunnel at the University of Notre Dame, a direct drag measurement system has been developed utilizing a linear force balance. In combination with indirect drag measurements taken at NASA Langley's Curved Duct Test Rig, measurement confidence has been established. This paper will detail the capabilities of the direct drag measurement system at Notre Dame and explore its intended future use in developing new liner technologies.

FRIDAY MORNING, 27 MAY 2016 SALON H, 8:00 A.M. TO 12:15 P.M.

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics II

Michael B. Muhlestein, Chair
Mechanical Engineering, University of Texas at Austin, 3201 Duval Rd. #928, Austin, TX 78759

Contributed Papers

8:00

5aPA1. Acoustic hysteresis of varying cavity length of a bottle-shaped thermoacoustic prime mover with a neck to cavity ratio of 1:10. Emily Jensen and Bonnie Andersen (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84058, em.jensen88@gmail.com)

A previous study showed hysteresis in transition regions to overtones of bottle-shaped thermoacoustic prime movers with a neck to cavity diameter ratio of 1:2.4 while varying the cavity length. Hysteresis regions were studied with a neck to cavity diameter ratio of 1:10. The device consisted of a neck (5.15 cm long, 0.75" ID) with a heating element around it and a cavity (ID 3.75") with a sliding piston, allowing the cavity length to be varied up to 38 cm long. Copper mesh was used for the heat exchangers and were located about 30{\%} away from the top of the neck and 16 mg of steel wool served as the stack. A pressure sensor was connected to the center of the piston to measure acoustic pressure at the bottom of the cavity. Acoustic pressure, frequency, and hot and cold temperatures were recorded while both increasing and decreasing the cavity length from about 2 to 38 cm in increments of 0.2 cm over time intervals of 20 s with an input power of 10.5 W. Three transitions to overtones occurred at 6.8, 15.2, and 27.4 cm while pulling the piston out and at about 25.2, 15.6, and 6.0 cm while pushing the piston in. Frequencies and transition regions agreed with expected values. Both hot temperature and acoustic pressure increased during transition regions. This could be caused from multiple acoustic waves being produced.

8:15

5aPA2. Experimental demonstration of mitigating flame-sustained thermoacoustic oscillations by using an electrical heater. Dan Zhao and Xinyan Li (Aerosp. Eng. Div., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore 639798, Singapore, zhaodan@ntu.edu.sg)

In this work, experimental investigation is conducted to mitigate flame-sustained thermoacoustic instability occurred in a conventional Rijke-type and bifurcating thermoacoustic systems by using an electrical heater. A premixed methane-burned flame is confined in the bottom part of the Rijke-type and bifurcating thermoacoustic systems to produce self-sustained thermoacoustic oscillations. As the electrical heater is placed downstream of the flame, it acts like a sound absorber and can even completely stabilize the unstable thermoacoustic system. The damping performance of the heater depends strongly on its axial location and output power. To further validate our findings, experimental investigation of mitigating thermoacoustic oscillations in a bifurcating unstable system by using a heater is performed. It has a mother tube with a premixed methane-burned flame confined. The mother tube splits into two bifurcating branches with an angle of α. A temperature-controllable electrical heater (TCH) is enclosed in one of the bifurcating branches. It is shown that the bifurcating system is associated with thermoacoustic oscillations at around 190 Hz with a sound pressure level of 130 dB. However, when the heater placing at 15 cm away from the open end is actuated, sound pressure level is found to be reduced by approximately 50 dB.

8:30

5aPA3. Using helium as the working fluid in thermoacoustic engines. Nathaniel Wells and Bonnie Andersen (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84058, nateswells@gmail.com)

The efficiency of thermoacoustic engines can be improved by using helium as the working fluid because it reduces viscous losses, has a higher thermal conductivity and speed of sound. The engine in this study has a bottle-shaped resonator. The neck consists of a brass cylinder, closed at the top end and a copper cylinder, open at both ends, with copper mesh screens heat exchangers between them (ID of 1.91 cm and total length of 5.24 cm). A small amount of steel wool (20 mg) functions as the stack. The neck opens into an aluminum cavity (10 cm long with an ID of 4.13 cm). A combination of two types of heat-shrink tubing and Teflon were used to connect the brass and copper pieces. The engine was evacuated of air and backfilled with helium as much as the setup would tolerate. Using an input power of 14.8 W over intervals of 0.5–3 h, it was observed that the frequency decreased in time, indicating that the helium was leaking out slowly. From the frequency data, the volume fraction of helium was calculated, indicating that the engine was able to achieve 64% volume fraction of helium and decreased to 6%. The intensities of the sound over this range of volume fractions averaged at 155 W/m^2 compared with air at an average of 118 W/m^2.

8:45

5aPA4. Entrainment of two thermoacoustic engines. Orest G. Symko, Myra Flitcroft, Brenna Gillman, Ivan Rodriguez, and Cedric C. Wilson (Phys. & Astronomy, Univ. of Utah, 621 S. 1100 E., Salt Lake City, UT, wilson.cedric.c@gmail.com)

In order to build more powerful sources of sound for energy conversion, the synchronization of two thermoacoustic heat engines has been studied. Experiments were performed on engines in the acoustic frequency range of 2.6 kHz and also on very small engines in the ultrasonic range of 24 kHz. In both cases, the engines were mounted on a cylindrical cavity, and they were coupled mainly by the acoustic field in the surrounding air at one atmosphere. They were driven by heaters of resistance wire in contact with the hot heat exchanger. At a specific coupling between the 2.6 kHz, engines' synchronization occurred, and also for the 24 kHz devices; frequency pulling in each pair of engines led to a common frequency in each set, i.e., in-phase synchronization. The strength of the synchronization was determined as a function of detuning of engines. Mutual entrainment was observed at the onset of oscillations and this is attributed to drive by fluctuations. Moreover, as a result of synchronization, the critical temperature gradient for onset of oscillations was reduced from that of the individual values. Delays between two oscillators in the start up led to quenching of the generated sound output (oscillation "death" of Rayleigh).

9:00

5aPA5. Irregular reflection of weak acoustic shock pulses on rigid boundaries. Desjouy Cyril, Sébastien Ollivier (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Univ. Lyon 1, 36 av Guy de Collongue, Ecully 69134, France, sebastien.ollivier@univ-lyon1.fr), Olivier Marsden, Didier Dragna, Maria Karzova, and Philippe Blanc-Benon

The reflection of weak shockwaves on rigid boundaries at grazing angles was studied in different geometrical configurations. It was shown that even in the case of weak shocks, irregular reflection of N-waves can lead to the formation of a three shocks pattern with a Mach stem, a triple point above the rigid surface and an angle reflection which differs from the incident one. Experiments were done using spark generated spherical shock waves with peak pressure lower than 10 kPa. Reflection patterns of shocks were obtained using a Schlieren visualization setup. Both regular and irregular regimes of reflection were observed. Numerical simulations of the nonlinear propagation were also performed. They were based on the high-order finite difference solution of the two dimensional Navier-Stokes equations. Optical measurements were compared with the results of simulations. Both experimental and numerical results showed the growth of the Mach stem with the distance of propagation. In the case of a randomly or periodically rough surface, the Mach stem is shorter, and nonlinear interactions between reflected waves occurs. [This work was supported by the Labex Centre Lyonnais d'Acoustique of Université de Lyon, operated by the French National Research Agency (ANR-10-LABX-0060/ANR-11-IDEX-0007).]

9:15

5aPA6. Derivative skewness values of shock-containing noise waveforms. Brent O. Reichman, Kent Gee (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), Michael James, Alexandria Salton (Blue Ridge Res. and Consulting, LLC, Asheville, NC), and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

Acoustic shocks present in military aircraft noise are often referred to as crackle, a component of jet noise that can significantly alter perception and annoyance. Quantifying the shock content within a waveform is important for gauging possible effects that these shocks may have. In the past, the derivative skewness has been used to quantify the steepening of waveforms throughout propagation, representing average behavior throughout the waveform. Though past use has been mostly qualitative, recent work has given physical meaning to derivative skewness values of sinusoids and provided recommendations on sampling rate and thresholds that can indicate significant shocks within the waveform. Using these recommendations, the derivative skewness of high-performance military aircraft noise is estimated for various locations and engine conditions. Derivative skewness values are compared with shock counting schemes based on various criteria. The comparisons show a strong relationship between high derivative skewness values and the strength and number of shocks within a waveform.

9:30

5aPA7. Comparison of rocket launch data to propagation model results. John Noble (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, john.m.noble.civ@mail.mil)

NASA's Wallops Flight Facility launched medium lift rockets for experimental and space station resupply missions during 2013 and 2014. These launches have been a great opportunity to use the rocket-generated infrasound as a repeatable source to study long range propagation. Data from the US Array was used to compare the received amplitude from the rocket launches to a propagation model prediction using realistic atmospheric profiles for ranges out to 500 km from the launch point. The US Array was a distribution of infrasound and seismic sensors which range in a north/south strip across the United States and would periodically relocate further east over time. The results of this comparison will show how well the modeling was able to track the behavior in the measurements.

9:45

5aPA8. Quadspectral nonlinearity analysis of military jet aircraft noise waveforms. Kyle G. Miller (Phys. and Astronomy, Brigham Young Univ., 323 East 1910 South, Orem, UT 84058, kglenmiller@gmail.com), Kent L. Gee, Brent O. Reichman, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Understanding the impact of jet noise can be improved by quantifying the nonlinearity in a signal with a single-microphone measurement. Based on the quadspectral Morfey-Howell indicator, a nonlinearity gain factor called ν_N has been derived from an ensemble-averaged, frequency-domain version of the generalized Burgers equation [Miller *et al.*, AIP Conf. Proc. **1685**, 090003 (2015)]. This gain factor gives a quantitative expression for the change in sound pressure level spectrum over distance. Past results show that ν_N accurately characterizes nonlinear evolution of waves in simulation

and model-scale jet data. Here, noise waveforms from a high-performance military jet aircraft are characterized using the ν_N indicator; results are compared with those from other indicators that have been used previously (e.g., derivative skewness, time-waveform steepening factor, etc.). Far field results show that the nonlinear gains at high frequencies (>1 kHz) tend to balance the absorption losses, thus establishing the characteristic $1/f^2$ spectral slope. Additionally, trends over angle, distance, and engine condition are explored. [Work supported by the AFRL SBIR program.]

10:00–10:15 Break

10:15

5aPA9. Modeling of the pressure distribution, acoustic streaming, and formation of bubble structures in an ultrasonic horn reactor. Yezaz A. Gadi Man and Francisco J. Trujillo (School of Chemical Eng., The Univ. of New South Wales, F10 Chemical Sci. Bldg., Kensington Campus, Sydney, NSW 2052, Australia, y.gadiman@unsw.edu.au)

A framework is developed to simulate the acoustic streaming and the formation of conical bubble structures (CBS) in ultrasonic horn reactors. In these reactors, acoustic pressure waves propagate from the vibrating solid horn to the liquid. On its movements away from the horn, sound is attenuated producing acoustic streaming, which is exacerbated by the presence of inertial acoustic bubbles that diminish the pressure amplitude many fold very near to the acoustic source. This is a complex Multiphysics phenomena occurring at different time and length scales. Inertial bubbles, which have sizes of the order of few micrometers, oscillate, grow, and collapse as function of time within the period of the acoustic wave while acoustic streaming is a time independent hydrodynamic movement of the liquid occurring through the totality of the reactor. The four main phenomena, solid vibration, acoustic pressure, streaming and bubble generation and distribution, are simulated in COMSOL Multiphysics in a stepwise fashion. The first two phenomena are coupled and solved in frequency domain, and the resultant acoustic pressure forces are applied to solve turbulent streaming and a transport equation in time-independent mode. Steps are repeated sequentially until a good agreement is found with experimental CBS, power, and streaming.

10:30

5aPA10. Quasi-emulsion between water and cavitation bubble cloud in an ultrasonic field. Lixin Bai, Weijun Lin, Jingjun Deng, Chao Li, Delong Xu, Pengfei Wu, and Lishuo Chen (Inst. of Acoust., Chinese Acad. of Sci., No. 21, Bei-Si-huan-Xi Rd., Beijing 100190, China, blx@mail.ioa.ac.cn)

The cavitation bubble distribution (cavitation structure) is spatially inhomogeneous in ultrasonic field. A quasi-emulsion phenomenon of cavitation structure is reported in this paper. The inception process of cavitation bubble cloud in a thin liquid layer (the thin liquid layer is trapped between a radiating surface and a hard reflector) was investigated experimentally with high-speed photography. It is revealed that cavitation bubble cloud can be considered as a uniform fluid (cloud), and water without cavitation can be considered as another uniform fluid (water). The conversion from cloud-in-water emulsion to water-in-cloud emulsion occurs with the increase in the number of bubbles. The formation and stability of cloud-water emulsions is analyzed theoretically. It is found that surface tension of cavitation bubble cloud played a leading role. Findings of this research proved that cavitation bubble clusters can be considered and investigated as a whole. [This work was supported by the National Natural Science Foundation of China (Grant No. 11174315).]

10:45

5aPA11. Dynamics of cavitating layer on a surface of quasi-empty cavity in a real heterogeneous liquid. Valeriy Kedrinskiy (Physical HydroDynam., Lavrentyev Inst. of HydroDynam., Russian Acad. of Sci., Lavrentyev Prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru) and Ekaterina S. Bolshakova (Physical Faculty, Novosibirsk State Univ., Novosibirsk, Russian Federation)

The state dynamics of a heterogeneous medium surrounding the spherical cavity is considered, using the multi-phase mathematical model with an incompressible liquid component. The heterogeneous medium contains 1.5-micron bubbles with a density of 10^6 cm^{-3}. At t = 0, the pressure in the cavity is sharply decreased up to p (0) values forming instantly the dynamically changing decompression wave in the surrounding medium. Numerical analysis allowed to find the significant influence of p (0) values on the dynamics of the cavitating layer on the cavity surface. When p (0) = 10^{-2} MPa, approximately 20 pulsations of bubble radii and their internal pressure are observed during the collapse time of 1 cm cavity (T is 0.9 ms). Most of them reach only its initial state with a pressure about 0.1 MPa. But at approaching to the time moment close to T value, a few pressure pulsations demonstrate the cumulative effect of over-compression up to the pressures of 0.6, 1.8, and 3 (in log-scale). The letter means that cavitating boundary layer accumulates the energy of high density. When the p (0) = 10^{-4} MPa, only one pulsation synchronous with the cavity collapse as well as the over-compression effect are observed. [Work supported by RFBR, grant 15- 05-03336.]

11:00

5aPA12. Ultrasound directed self-assembly of user-specified patterns of nanoparticles dispersed in a fluid medium. John Greenhall, Fernando Guevara Vasquez, and Bart Raeymaekers (Mech. Eng., Univ. of Utah, 1495 e 100 s, Salt Lake City, UT 84112, john.greenhall@utah.edu)

We employ an ultrasound wave field generated by one or more ultrasound transducers to organize large quantities of nanoparticles dispersed in a fluid medium into two-dimensional user-specified patterns. To accomplish this, we theoretically derive a direct method of calculating the ultrasound transducer parameters required to assemble a user-specified pattern of nanoparticles. The computation relates the ultrasound wave field and the force acting on the nanoparticles to the ultrasound transducer parameters by solving a constrained optimization problem. We experimentally demonstrate this method for carbon nanoparticles in a water reservoir and observe good agreement between experiment and theory. This method works for any simply-closed fluid reservoir geometry and any arrangement of ultrasound transducers, and it enables using ultrasound directed self-assembly as a scalable fabrication technique that may enable a myriad of engineering applications, including fabricating engineered materials with patterns of nanoscale inclusions.

11:15

5aPA13. Real-time polymerization monitoring of a thermosetting resin around its glassy transition temperature. Nacef Ghodhbani, Pierre Marechal, and Hugues Duflo (LOMC, UMR 6294 CNRS, Université du Havre, 75 rue Bellot, Le Havre 76600, France, nacef.ghodhbani@univ-lehavre.fr)

Real-time ultrasonic monitoring is investigated to quantify changes in physical and mechanical properties during the manufacture of composite structures. In this context, an experimental transmission was developed with the aim to characterize a high temperature polymerization reaction and post-curing properties using an ultrasonic method. First, the monitoring of ultrasonic parameters of a thermosetting resin is carried out in an isothermal polymerization process at 160°C. During this curing, the resin is changing from its initial viscous liquid state to its final viscous solid state. Between those states, a glassy transition stage is observed, during which the physical properties are strongly changing, i.e., an increase of the ultrasonic velocity up to its steady value and a transient increase of the ultrasonic attenuation. Second, the ultrasonic inspection of the thermosetting resin is performed during a heating and cooling process to study the temperature sensitivity after curing. This type of characterization lead to identify the ultrasonic properties dependence before, during, and after the glassy transition temperature T_g. This study is composed of two complementary parts: the first is useful for the curing optimization, while the second one is fruitful for the post-processing characterization in a temperature range including the glassy transition temperature.

11:30

5aPA14. Yeast flocculation using acoustic agglomeration. Mark H. Holdhusen (Eng., Univ. of Wisconsin, Marathon County, 518 S 7th Ave., Wausau, WI 54401, mark.holdhusen@uwc.edu)

A major cause of haziness in beer is due to yeast suspended in the liquid. The most common methods used to flocculate the yeast and have it settle

out of the liquid solution are to use various chemicals, cold temperatures, and/or extended time. This research considers using acoustic agglomeration to cause the yeast to flocculate as a means to clarify beer. Acoustic agglomeration uses high intensity acoustic standing waves to clump fine particles together in order for them to become large enough to settle out of the fluid. In this work, an ultrasonic acoustic transducer will be implemented to achieve standing waves in the beer. In theory, these standing waves will cause the yeast particles to clump together and settle. The results from this approach will be compared to the traditional methods of yeast flocculation. This approach may lead to an increase in efficiency in clarifying beer without the use of undesirable chemicals. The preliminary work considered here is a proof of concept and will use visual inspection as a means of clarity comparison. Future work will use more in-depth laboratory analysis as a means of comparison of clarity.

11:45

5aPA15. Photoacoustic spectroscopy with SF_6, an optically thick greenhouse gas. Han Jung Park and Wittmann S. Murphy (Chemistry, Univ. of Tennessee at Chattanooga, 615 McCallie Ave., Chattanooga, TN 37403, hanjung-park@utc.edu)

Photoacoustic spectroscopy was used to test the photoacoustic properties of sulfur hexafluoride, an optically thick and a potent greenhouse gas. Detection of trace amounts of the gas was also implemented. The conditions in which the gas was tested, gas cell length, temperature, concentration, and power of the laser, were varied to determine their effect on the photoacoustic signal, and the ideal conditions to detect trace gas amounts. A detection limit of 2.86 ppb was determined for SF_6.

FRIDAY MORNING, 27 MAY 2016 — SALON D, 7:55 A.M. TO 12:00 NOON

Session 5aPP

Psychological and Physiological Acoustics: Spatial Hearing

Ewan A. Macpherson, Chair

National Centre for Audiology, Western University, 1201 Western Rd., Elborn College 2262, London, ON N6G 1H1, Canada

Chair's Introduction—7:55

Contributed Papers

8:00

5aPP1. Listener head motion can degrade spatial selective auditory attention. Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd., Elborn College 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca) and Blair K. Ellis (Health and Rehabilitation Sci. Graduate Program, Western Univ., London, ON, Canada)

Listeners can use spatial selective auditory attention (SSAA) to focus on one talker in a complex acoustic scene. Although the dynamic binaural cues generated by listener head motion improve sound localization, and might therefore enhance the perceptual differences between separated targets and distractors, the effect of head motion on SSAA is unknown. We measured listeners' ability to attend, with and without head motion, to a frontal target in the presence of two symmetrically separated distractors. During stationary trials, listeners visually fixated and oriented toward a target loudspeaker. On head-motion trials, listeners oscillated their heads at ~0.5 Hz with an amplitude of $\sim\pm40^\circ$ while continuously directing their gaze toward the target. On each trial, three equal-intensity sequences of four spoken digits were presented simultaneously as target and distractors. Listeners reported the target sequence heard. With distractors at $\pm22.5^\circ$, 86% of target digits were reported correctly without motion, but only 72% were reported correctly with head motion. Correspondingly, the percentage of distractor digits reported as targets increased from 11% to 23%. For widely separated distractors, there was no performance penalty for head motion. These results suggest that listeners cannot rapidly update the focus of their SSAA to compensate for head motion.

8:15

5aPP2. Restoring sensitivity to interaural timing differences in bilateral cochlear implant listeners using multi-electrode stimulation. Tanvi D. Thakkar, Alan Kan, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, tthakkar@wisc.edu)

Identification and discrimination of sound sources in complex auditory environments is facilitated for normal-hearing listeners from access to interaural time differences (ITDs). For patients fitted with bilateral cochlear implants (BiCI), binaural sensitivity is harder to achieve due to several factors, including asynchronous interaural processing. BiCI listeners have shown good ITD sensitivity with low rates of electrical stimulation; however, low-rate delivery of ITD cues is unrealistic for the high-rate pulsatile stimulation required to achieve good speech understanding. One solution is to present a mix of high- and low-rate stimulation on different electrodes, preserving both speech recognition and sound localization ability. In the present study, a binaural benefit was observed in a mixed-rate strategy under direct electrical stimulation: ITD sensitivity was measured in a two-alternative forced-choice discrimination task, using seven multi-electrode conditions. We hypothesized that by introducing low-rate ITDs at a few electrodes alongside high-rate ITDs at remaining electrodes provides sufficient ITD cues for sound localization. The present data suggests the binaural system of BiCI listeners can extract pertinent cues to achieve ITD sensitivity even when high-rate ITD information is presented at the majority of the cochlear locations. This lends to a possibility in implementation of ITD cues to current processing strategies.

8:30

5aPP3. Shared monaural and binaural temporal processing limits in bilateral cochlear implant listeners. Antje Ihlefeld (Biomedical Eng., New Jersey Inst. of Technol., 323 Martin Luther King Blvd., Fenster Hall, Rm. 645, Newark, NJ 07102, ihlefeld@njit.edu), Robert P. Carlyon (Cognition and Brain Sci. Unit, Medical Res. Council, Cambridge, United Kingdom), Alan Kan, Tyler H. Churchill, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI)

Bilaterally implanted cochlear implant users were tested on monaural rate discrimination and binaural interaural time difference (ITD) discrimination, as a function of pulse rate, to examine the hypothesis that deterioration in performance at high rates occurs for the two tasks due to a common neural basis. For the rate discrimination task, pulse trains were presented to one electrode, located in the apical, middle, or basal part of the array, and in either the left or the right ear. In each two-interval trial, the standard stimulus had a rate of 100, 200, 300, or 500 pulses-per-second and the signal stimulus had a rate 35% higher. For the ITD discrimination task, performance between pitch-matched electrode pairs was measured for the same standard rates as in the rate discrimination task, and with an ITD of $+/-$ 500 μs. Sensitivity (d') on both tasks decreased with increasing rate. Results show that ITD scores for different pairs of electrodes correlated with the lower of the rate discrimination scores for those two electrodes. Statistical analysis, which partialed out overall differences between listeners, electrodes, and rates, supports the hypothesis that monaural and binaural temporal processing limitations are at least partly due to a common mechanism.

8:45

5aPP4. The perception of reverberation is constrained by environmental statistics. James Traer and Josh H. McDermott (Brain and Cognit. Sci., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, jtraer@mit.edu)

Human sound recognition is robust to reverberation. We explored the hypothesis that this robustness is rooted in the ability to separate the contributions of a sound's source from that of reverberation. As the separation of source and filter from their convolution is inherently ill-posed, any such capacity should depend on prior assumptions about the nature of filter and/or source. We measured the distribution of real-world environmental impulse responses (IRs) and tested whether it constrains the ability of listeners to estimate source and filter from reverberant audio. We surveyed volunteers about the spaces they encountered during daily life, and measured IRs at each location. We found that the tails of real-world IRs decay exponentially, with decay rates consistently slower at low frequencies. We then synthesized IRs that were either faithful to, or deviated from, the observed distribution of real-world IRs. We assessed (a) the sense of reverberation conveyed by IRs, (b) discrimination of novel sound sources in reverberation, and (c) discrimination of IRs given only their convolution with sound sources. We found that human listeners can separately estimate the source and filter in reverberant conditions, but are strongly constrained by whether the filter conforms to the naturally occurring distribution.

9:00

5aPP5. Effect of visual and vestibular information on auditory space perception. Shuichi Sakamoto, Keishi Hanakago, Zhenglie Cui (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Wataru Teramoto (Faculty of Letters, Kumamoto Univ., Kumamoto, Japan), Yôiti Suzuki (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., Sendai, Japan), and Jiro Gyoba (Graduate School of Arts and Letters, Tohoku Univ., Sendai, Japan)

We investigated how auditory space would be represented during linear self-motion and visually-induced self-motion (vection) (Teramoto *et al.*, 2012, 2014). The previous studies indicated that the subjective coronal plane (SCP) was displaced during forward self-motion, while during backward motion in the case of vection. The present study investigated how auditory space perception would be altered when both visual and vestibular information were presented. A random-dot pattern simulating linear self-motion was used. At the beginning of the trial, the random-dot pattern started to move at a velocity of ± 0.1 m/s. When an observer perceived vection, the velocity was changed at an acceleration of ± 0.2, 0, and 0.4 m/s^2. At the same time, the observer was moved forward at an acceleration of 0.2 m/s^2 by using a linear-motor-driven chair. A short noise-burst was presented from a loudspeaker when the observer moved 2 m. The observers indicated the direction in which the sound was perceived relative to their coronal plane. The results showed that the direction of the SCP shift was different only when the acceleration of both visual and vestibular information was identical. These results suggest that auditory space distortion effect occurs closely related to the integration of vestibular and visual information.

9:15

5aPP6. Sound-source enumeration by hearing-impaired adults. Michael A. Akeroyd (MRC Inst. of Hearing Res., Nottingham NG7 2RD, United Kingdom, maa@ihr.mrc.ac.uk), William M. Whitmer, David McShefferty, and Graham Naylor (Scottish Section, MRC/CSO Inst. of Hearing Res., Glasgow, United Kingdom)

To help measure the veracity with which the auditory world is heard by hearing-impaired listeners, we studied the ability to count the number of discrete sources that are present. In two experiments, stimuli were newly selected 5-s samples of concatenated sentences from 1 to 7 locations in simulated rooms of varying reverberation, presented from a circular loudspeaker array. In the first experiment, listeners responded with the number of locations heard after each presentation. In the second experiment, listeners heard two presentations, an interval with n sources (one talker per location) and an interval with $n+1$ sources; listeners responded which interval had more sources. Results corroborate recent work showing the maximum number of identifiable sources is roughly four. Asymmetry in hearing impairment reduced the ability to enumerate locations. Various conditions of presentation were tested: headphone and aided results were not markedly different from unaided results, and reverberation had only a modest inflating effect on the perceived number of sources. The ramifications for complex listening as well as the potential relation to cortical representations of auditory space are discussed. [Work supported by the MRC (U135097131) and the Chief Scientist Office (Scotland).]

9:30

5aPP7. Evaluating performance of hearing-impaired listeners with a visually-guided hearing aid in an audio-visual word congruence task. Elin Roverud (Boston Univ., 712 Cincinnati St., Lafayette, IN 47901, emroverud@gmail.com), Virginia Best, Christine R. Mason, Timothy Streeter, and Gerald Kidd (Boston Univ., Boston, MA)

Hearing-impaired (HI) individuals typically experience greater difficulty listening selectively to a target talker in speech mixtures than do normal-hearing (NH) listeners. To assist HI listeners in these situations, the benefit of a visually guided hearing aid (VGHA)—highly directional amplification created by acoustic beamforming steered by eye gaze—is being evaluated. In past work with the VGHA, performance of NH listeners was assessed on an audio-visual word congruence task [e.g., Roverud *et al.*, ARO 2015]. The present study extends this work to HI listeners. Three spoken words are presented simultaneously under headphones at each of three source locations separated in azimuth. A single word is printed on a monitor at an angle corresponding to one of the locations, indicating the auditory target word location. After each stimulus, listeners indicate whether the auditory target matches the printed word with a YES-NO response, ignoring the two masker words. Listeners visually track the printed word, which moves location trial-to-trial with a predetermined probability. Participants' eye movements, measured by an eye tracker, steer the VGHA amplification beam. Performance is compared for the VGHA, stimuli processed with natural binaural information (KEMAR HRTFs), and a hybrid condition combining both types of information. [Work supported by NIDCD.]

9:45

5aPP8. Temporal integration of dynamic binaural events. G. Christopher Stecker, Julie M. Stecker, Anna C. Diedesch, Nathan C. Higgins, and Sandra Da Costa (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu)

Events occurring in different sensory dimensions (e.g., modalities) and within an interstimulus interval (ISI) of few hundred milliseconds typically

appear simultaneous. The temporal "window" of sensory integration can be estimated by increasing the ISI until events appear discrete. Here, we adapt the simultaneity-judgment paradigm to investigate the integration of binaural events within and across dimensions of interaural time and level difference (ITD and ILD). Noise bands (353.5–707 Hz) of 2s duration were presented over headphones to normal-hearing listeners. Each stimulus contained two 10-ms binaural-change events, during which the ITD or ILD changed by 500 μs or 6 dB. Events could occur in the same or different cue dimensions (ITD or ILD), but always agreed in direction (leftward or rightward). ISI varied over the range ±600 ms. Listeners indicated the number and direction of each perceived shift in the lateral image, e.g., one (left-to-right) at short ISI, or two (left-to-center, center-to-right) at long ISI. The threshold for reporting two events was shortest for pairs of ITD events, suggesting greater temporal fidelity for ITD change than for ILD change. Thresholds did not differ between ILD and cross-cue event pairs, suggesting no additional cost of cross-cue integration. [Work supported by NIH R01-DC011548.]

10:00–10:15 Break

10:15

5aPP9. Sensitivity to binaural cues beyond threshold as revealed by eye movements. Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St, Seattle, WA 98105, mwinn83@gmail.com) and Alan Kan (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI)

Binaural cues are paramount for sound localization along the azimuth. Studies on the perception of the critical binaural cues, interaural time and level differences (ITDs and ILDs, respectively), typically measure sensitivity at threshold using N-interval, forced choice paradigms. This approach gives little or no information regarding perceptual abilities beyond threshold. In this study, an anticipatory eye movement (AEM) paradigm (cf. McMurray, 2004) was used as a novel measure to study binaural cue sensitivity throughout a wide perceptual range. This paradigm is sensitive to gradient (rather than all-or-none) perception of auditory cues. Adults with normal hearing visually tracked the location of a ball that becomes hidden on a computer screen, and anticipated its motion and reappearance via eye movements guided by binaural cues. The auditory stimuli were 4.8-s 1/3-octave narrowband noises centered at different frequencies, and contained an ILD or ITD change, indicating the impending motion of the ball. Greater cue levels elicited systematically quicker and more accurate saccades. ILD-driven AEMs were elicited for all noise center frequencies, with greater sensitivity for higher frequency noises, consistent with ecological significance. Eye movements in this study suggest variation in binaural cue perception beyond threshold that is gradient in terms of latency and accuracy.

10:30

5aPP10. The role of early and late reflections on spatial release from masking. Nirmal Kumar Srinivasan, Meghan M. Stansell, Rachel E. Ellinger, Kasey M. Jakien, Sean D. Kampel, and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, nirmal.srinivasan@va.gov)

It is well documented that older listeners have more difficulty understanding speech in complex listening environments than do younger listeners. Early reflections (occurring <50 ms following the direct sound) have been linked to improved speech intelligibility (Lochner and Burger, 1964), while later-arriving reverberant sound has been shown to limit speech understanding (Knudsen, 1929). However, we do not know how spatial release from masking (SRM) is affected by early and late reflections or how age and hearing loss interacts with the relative influences of each. SRM in two simulated reverberant environments was measured for listeners varying in age and hearing thresholds under three different signal processing conditions: (1) early reflections alone, (2) late reflections alone, and (3) all reflections. Results indicated that though all listeners performed better when only early reflections were present, the older hearing-impaired listeners benefited the most from the absence of late reflections. Effects of age and hearing loss on performance and SRM under these three different signal processing conditions will be discussed. [Work supported by NIH R01 DC011828.]

10:45

5aPP11. Discrimination and streaming of speech sounds based on differences in lateralization in the horizontal and median planes. Marion David and Andrew Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, david602@umn.edu)

Understanding speech in complex backgrounds relies on our ability to perceptually organize competing voices into streams. Differences in fundamental frequency (F0) between voiced sounds are known to enhance stream segregation; less is known about the perceptual organization of unvoiced sounds such as fricative consonants. We showed previously that natural consonant-vowel (CV) pairs can be segregated based on F0 differences, despite lacking F0 cues in the fricative part. This study also used CVs, filtered by head-related impulse responses (HRIR) to simulate different positions in the horizontal and median planes. In the median plane, cues are limited to high frequencies, and so should affect fricatives more than vowels. Both discrimination, using a three-interval forced-choice task, and streaming, using a within- or across-stream repetition-detection task, were tested. The CV pairs were either held constant during a trial, or varied randomly. In the constant condition, any difference in spectrum would indicate a change in location along the median plane; in the variable condition, such differences would require listeners to extract spectral regularities from across widely varying spectra of the speech stimuli. Preliminary results suggest that discrimination and streaming are more challenging in the median than in the horizontal plane. [Work supported by NIH grant R01DC012262.]

11:00

5aPP12. Acoustic cues for determining head position in sound source localization when listeners move. William Yost (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Recently, Yost *et al*. [J. Acoust. Soc. Am. **40**, 3293–3310 (2015)] showed in experiments involving rotating listeners that everyday world-centric sound source localization involves two sources of information: information about the auditory spatial cues and information about the position of the head. In this presentation, we will describe experiments in which sound has the potential to provide information about the position of the head, allowing for world-centric sound source location. We use conditions in which sound rotates along an azimuth circular array of loudspeakers as listeners also rotate in the azimuth plane at constant velocity with their eyes open or closed. The rotation conditions and the resulting perceptions of sound rotation will be described along with how those perceptions are altered when other sounds are present. [Research supported by the Air Force Office of Scientific Research, AFOSR.]

11:15

5aPP13. Binaurally integrated cross-correlation/auto-correlation mechanism (BICAM). Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

A new precedence effect model is described that can use a binaural signal to robustly localize a sound source in the presence of multiple reflections. The model also extracts the delays (compared to the direct sound source) and lateral positions of each of the distinct reflections. A second-layer cross-correlation algorithm is introduced on top of a first layer auto-correlation/cross-correlation mechanism to determine the interaural time difference (ITD) of the direct sound source component. The ITD is then used to time align two auto-correlation functions obtained from the left and right ear signals to gather information about the reflections and form a binaural activity pattern. The model is able to simulate psychoacoustic lateralization results for a simulated direct sound source and a single reflection also for cases where the reflection exceeds the intensity of the direct sound (Haas Effect). Using head-related transfer functions to spatialize the sound sources, the model can accurately localize a speech signal in the presence of two or more early side reflections and late reverberation. The model can handle reverberation times of 2 s and above. [This material is based upon work supported by the National Science Foundation under Grant No. 1320059.]

11:30

5aPP14. Squelch of room effects in everyday conversation. Aimee Shore, William M. Hartmann, Brad Rakerd (Michigan State Univ., Phys. Astronomy, 567 Wilson Rd., East Lansing, MI 48824, shoreaim@msu.edu), Gregory M. Ellis, and Pavel Zahorik (Univ. of Louisville, Louisville, KY)

When a conversation is recorded and then played back, listeners are aware of effects of the room that are not perceived in face-to-face conversation. The room effects, which are physically present, are said to be "squelched" under face-to-face conditions. Room effects include reverberation and coloration caused by reflections. The squelch effect has been attributed to the binaural nature of natural listening, based on binaural experiments with headphones [W. Koenig, J. Acoust. Soc. Am., {22}, 61–62 (1950)]. We report experiments on binaural and diotic listening to recordings of speech made at conversational distances in a room with normal frequency dependence of reverberation and a direct to reverberant power ratio between 1/2 and 1/3 at speech fundamental frequencies. Recordings were made with and without a head between the microphones. Listeners ranked the recordings in order of increasing perceived room effects. The data revealed a strong effect of distance, a weak effect of head diffraction, and an advantage for binaural listening somewhat smaller than the advantage of a factor of 2 in direct power. For a few listeners, binaural listening enhanced the room effects. The latter listeners apparently found that binaural differences produced especially prominent spatial effects. [Work supported by the AFOSR.]

FRIDAY MORNING, 27 MAY 2016 — SALON B/C, 8:00 A.M. TO 9:50 A.M.

Session 5aSCa

Speech Communication: Phonetics of Under-Documented Languages I

Amanda L. Miller, Cochair
Linguistics, The Ohio State University, 222 Oxley Hall, 1712 Neil Avenue, Columbus, OH 43210-1298

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Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

Chair's Introduction—8:00

Invited Papers

8:05

5aSCa1. Under-documented languages expand phonetic typology. Ian Maddieson (Dept. of Linguist, UNM, Univ. of New Mexico, MSC03-2130, Albuquerque, NM 87131-0001, ianm@berkeley.edu)

The set of speech sounds known to be used in human languages continues to grow ever larger as more information becomes available on previously under-documented languages. In addition, the range of contrastive distinctions known to be employed continues to be enlarged. A brief survey of categories of sounds that have been added to phonetic typology as a result of work on such languages will be presented, followed by exemplification of the specific case of Yélî Dnye (ISO 693 yle). This language, spoken on Rossel Island, Papua New Guinea by about 3000 people, has large contrastive inventories of both consonants and vowels (58 consonants, 34 vowels). It is the only language known to include sets of doubly articulated labial-alveolar and labial-postalveolar plosives and nasals in its inventory (in addition to the more widespread category of labial-velars). Moreover, the stops contrast plain, prenasalized and nasally released categories, and some of them occur distinctively palatalized. Thus, there are at least nine consonant types not known from any other language. Yélî Dnye is also the only language known to have a contrast between oral and nasalized vowels following nasally released stops. Our knowledge of this language therefore enlarges our perspective on how human languages may differ.

8:30

5aSCa2. Acoustic realization of a distinctive, frequent glottal stop: The Arapaho example. Doug H. Whalen (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, whalen@haskins.yale.edu), Christian DiCanio (Haskins Labs., Buffalo, NY), Christopher Geissler, and Hannah King (Haskins Labs., New Haven, CT)

Complete closure of the glottis is typically treated as the canonical realization of glottal stop, but it has instead been found to be "quite unusual" in running speech. However, such evidence comes mostly from English, with non-phonemic glottal stops. How do glottal stops vary in a language where they are common and contrastive, as in Arapaho (ISO 639 arp)? Does distinctive and frequent use of

a glottal stop lead to more canonical productions? Moreover, glottalization is often used to mark prosodic boundaries; Are Arapaho phonemic glottal stops affected by boundary position? Glottal stops in an Arapaho corpus were classified on a scale of lenition and examined for duration, relative intensity, harmonics-to-noise ratio (HNR), and F0. Results show that glottal stops were seldom realized as a stop (only 25%) but instead mostly as glottalization. HNR was lower in glottalization than in adjacent vowels. Word-final glottal stops were more often realized with full closure than word-internal ones. The rarity of full glottal stops in English is also reflected in Arapaho: Greater use of these stops does not result primarily in canonical stop realizations. Moreover, glottal stop realization varies prosodically; thus, glottalization as a prosodic feature is not restricted to non-phonemic glottal stops.

8:50

5aSCa3. On the possible origin of voiceless implosives: Hints from Ese'eja (Takanan). Ddiier Demolin (ILPGA, LPP sorbonne nouvelle, 19 rue des Bernardins, Paris 75005, France, ddemolin@univ-paris3.fr) and Marine Wuillermet (Linguist, Radboud Universitei, Nijmegen, Netherlands)

Voiceless implosives are reported in a few languages of Africa, Mesoamerica, and Amazonia. Ese'eja (Takanan) has bilabial [ɓ̥] and [ɗ̥] alveolar voiceless implosives in its phonemic inventory (Vuillermet 2006). These sounds are realized with a complete closure of the vocal folds, a lowering of the larynx during the glottal closure associated with a lowering of the pressure (Po) inside the vocal tract behind the labial or alveolar closures. This is followed by a rapid larynx rising. The acoustic characteristics are: a period of silence and a short prevoicing preceding a strong final burst. There are two possibilities to explain the origin of voiceless implosives in Ese'eja. The first is that they are the consequence of the devoicing of a voiced implosive. The second is that they are due to the combination of a glottal closure and a voiced stop. The lowering of the glottis and Po is anticipating the articulation of a following low or back vowel during the glottal closure. Voiceless implosives have been described as preglottalized stops in other languages. (Dimmendaal 1986) emphasizes that in Lendu there is a possible auditory confusion between voiceless implosives and preglottalized stops. Aerodynamic and acoustic measurements confirm this hypothesis for Ese'eja.

9:10

5aSCa4. Interaction of pitch and phonation in Louma Oeshi. James Gruber (Dept of Linguist, Reed College, 3203 SE Woodstock Blvd., Portland, OR 97202-8199, james.gruber@reed.edu) and Sigrid Lew (Graduate Linguist Dept., Payap Univ., Chiang Mai, Thailand)

Louma Oeshi is an Akoid (Tibeto-Burman) language of Laos for which acoustic characteristics are undocumented with the exception of preliminary work by the present authors. This study focuses on the phonetic properties associated with Oeshi's three tones (high, mid, and low) and two registers (Tense, Lax), which fully intersect to yield a six-way suprasegmental contrast. Eight speakers were recorded in Phongsali Province, Lao PDR. Each spoke a 100 token word list, for which 30 tokens were repeated in a carrier sentence placing the token between high and mid-toned lax words. Decile measures of F0 and an array of measures reflecting phonation types (H1-H2, H1-A1, H1-A3, HNR, and SHR) were performed in Voicesauce (Shue *et al*. 2011) to capture dynamic values over the syllable duration. Our findings show a reliable three-way F0 contrast between tones and a single interaction with register such that High Tense words consistently fall (tones are otherwise level). Acoustic correlates of the Tense~Lax distinction are less clear. The picture that emerges is one where Tense register has variable phonetic manifestations — preglottalization of onsets, vocalic creaky voice, or a glottal stop coda — and the seemingly inconsistent acoustic results reflect these variable articulatory timing strategies.

9:30

5aSCa5. Investigating phonetic bases of sound patterns in Mangetti Dune !Xung: Contributions of high frame rate ultrasound to the description of endangered languages. Amanda L. Miller (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, miller.5592@osu.edu)

High frame rate lingual ultrasound methods (Miller and Finch 2011) allow investigations of consonant and vowel kinematics outside of the laboratory. I summarize the results of a number of studies that investigate place and manner of articulation in consonants produced with the pulmonic and lingual airstream mechanisms in the endangered Namibian language Mangetti Dune !Xung. Results show that clicks, like dorsal stops, exhibit different postures of the posterior part of the tongue when they precede [i] and [ɑ]. The tongue dorsum and root are retracted in the production of all four coronal clicks when they precede [ɑ], but differ in their postures when they precede [i]. Further, tongue dorsum and root postures are less variable within click types before [ɑ], than they are preceding [i]. Clicks also differ in the timing of the anterior and posterior releases, resulting in different constrictions being adjacent to following vowels, thus leading to different co-articulation patterns. Timing patterns of the two releases also contribute to our understanding of a diachronic sound change from an abrupt palatal click to a fricated post-alveolar click in this family. This body of research illustrates how high frame rate ultrasound contributes to descriptions of sound patterns in under-described languages.

Session 5aSCb

Speech Communication: Phonetics of Under-Documented Languages II (Poster Session)

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All posters will be on display and all authors will be at their posters from 10:05 a.m. to 12:00 noon.

Contributed Papers

5aSCb1. Acoustic phonetic study of phonemes and tonemes of spoken Punjabi language. Shyam S. Agrawal, Shweta Bansal, Shambhu Sharan, and Amritpal Singh (College of Eng., KIIT, Sohna Rd., Near Bhondsi, Gurgaon, Haryana 122102, India, dr.shyamsagrawal@gmail.com)

Punjabi language is one of the important languages among 22 official languages in India. It is spoken by about 105 million people in India. The present paper describes a study and results of detailed acoustic analysis of vowels, consonantal phonemes and tonemes as spoken by the speakers of Malwai dialect of Punjabi language. A database of 1500 words containing all the phonemes and tonemes, selected from a text corpus of 300, 000 words were used for the study. These words were recorded and segmented by using signal processing tools to analyze the samples of speech. Fundamental frequency, first three formants, and bandwidths for nasal and non-nasal vowels were measured. For the study of consonants, duration of sub-phonemic segments such as occlusion, burst, and VOT have been computed and compared. Special features of tonemes have been studied and compared with non-tonal phonemic segments. Tones are fully phonemic and words with similar spellings are distinguished by varying tones- low, mid, and high and corresponding accent marks. It has been observed that intonation plays a significant role in the discrimination of phonemes and tonemes. These results can be used to create PLS and phonetic dictionary of Punjabi speech.

5aSCb2. Consonant-tone interactions in Gengbe. Samson Lotven and Kelly Berkson (Dept. of Lingustics, Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

Gengbe is an under-documented and understudied Gbe language spoken as a lingua franca in Southern Togo and Benin. Few resources for Gengbe exist, especially in the domain of empirical acoustic phonetic research. To this end, we present an overview of its consonant, vowel, and tonal inventories. Of particular interest is the fact that its two register tones, (L)ow and (H)igh, show systematic phonological variation based on the voicing of onset consonants. Like many other tone languages in Africa and beyond, voiced obstruents act as so-called "depressor consonants," triggering lower f0 on subsequent vowels than do their voiceless counterparts. This lowering is phonologized in many H tone contexts, resulting in a Rising tone in many (but not all) morphophonological environments. The distinction is phonetically present in low tone as well, where it perseverates across the entire vowel and results in a lower register L (by approx. 20 Hz). We survey the phonological environments where such lowering effects are realized, and probe the interaction between obstruent voicing, tone, vowel height, and nasality via instrumental acoustic analysis.

5aSCb3. Pronunciation change in SENĆOŦEN: A acoustic study of /k, k^w, k^w', q, q', q^w q^w'/ across generations of speakers. Sonya Bird (Linguist Dept., Univ. of Victoria, PO Box 1700 STN CSC, Victoria, BC V8W 2Y2, Canada, sbird@uvic.ca)

This paper presents an acoustic study of /k, k^w, k^w', q, q', q^w q^w'/ in SENĆOŦEN, a dialect of North Straits Salish. The "K series"—as these sounds are collectively called—are of particular concern among speakers, the perception being that both the uvular ~ velar and the plain ~ ejective contrasts may be disappearing. To understand how these contrasts are currently being realized, 12 speakers (3 speakers x 4 generations) were recorded pronouncing 20 isolated words (7 consonants x 2–3 words). Based on auditory impression and preliminary acoustic analysis (PLACE: spectral composition of bursts/frication; formant transitions into adjacent vowels; VOICING: VOT; jitter; shimmer; amplitude rise time), the velar ~ uvular contrast is indeed less consistent among younger speakers than among their elders; there is also a lot of variability in the plain ~ ejective contrast, among all speakers. In addition, the plain ~ ejective contrast interacts with the velar ~ uvular contrast for some speakers: /k kw/ and /q qw/ are merged and relatively fronted ([k kw]); /kw'/ and /qw'/ are merged and relatively backed ([qw']). These realizations likely reflect a combination articulatory/aerodynamic considerations and orthographic influences (e.g., /kw'/ = 'Q'; /qw/ = 'Ḱ').

5aSCb4. A preliminary acoustic analysis of vowels and obstruents in San Juan Quiahije Chatino. Jacob Heredos, Kaitlynn Milvert, and Hilaria Cruz (Indiana Univ., 1021 E 3rd St., Memorial Hall 322 E (Linguistics), Bloomington, IN 47405, jheredos@indiana.edu)

San Juan Quiahije (SJQ) Chatino is a language of the Zapotecan branch of the Otomanguean family, spoken by approximately 4000 people in the state of Oaxaca in southern Mexico. While acoustic analyses of other varieties of Chatino exist, empirical acoustic data related to the phonetic inventory of SJQ Chatino is slim to nonexistent. As such, the current work presents a preliminary investigation of the stop consonants and vowels of SJQ Chatino. Data are from one female native speaker of SJQ and include words produced in isolation as well as in running speech. F1 and F2 values are used to plot the five oral and four nasal vowels of SJQ. With regard to the consonant inventory, contrastive voicing has been lost in at least one variety of Chatino and is marginal in a number of other varieties. Our data confirm that SJQ retains the contrast in coronal stops in at least some contexts. Voice onset times are reported, and negative values are found after initial /n/.

5aSCb5. Consonantal timing in Eastern Armenian. Knar Hovakimyan (Linguist, Reed College, #1328, 3203 SE Woodstock Blvd., Portland, OR 97202, knhovakim@reed.edu)

Eastern Armenian is an under-documented language, particularly in the field of phonetics. The language treats onset and coda consonant clusters asymmetrically, breaking up onset clusters with a vowel, but not coda clusters. This study focuses on consonantal timing in word-initial and word-final consonant sequences in Armenian. Five speakers were recorded reading 50 pairs of words with representative consonant clusters and equivalent words without clusters. Results indicate that singleton consonants behave differently than complex consonants depending on their location within a word. Namely, the vowel-adjacent consonant in a complex coda is longer in duration than an equivalent singleton consonant, whereas in onsets the duration of consonants is stable regardless of whether or not there is a preceding consonant. This corresponds to a greater degree of overlap in codas than in onsets. This asymmetrical treatment is motivated by the need to preserve phonetic cues for distinguishing consonants. The 30-consonant inventory is greatly limited by neutralizations in coda position, but all consonantal contrasts are maintained in onset clusters. Since there are more consonants to be distinguished in onset position, stronger cues are required to identify them and so less overlap is tolerated.

5aSCb6. Voicing contrast in Ruwund. Didier Demolin (LPP-ILPGA, Université Sorbonne nouvelle, 19 rue des Bernardins, Paris 75005, France, ddemolin@univ-paris3.fr)

Ruwund has a set of voiceless and voiced stops in its phonemic inventory. Length measurements reveal that voiceless stops are realized with duration comparable to voiced stops. There is also a positive VOT varying between bilabial, alveolar, and velar places of articulation. Voiced stops are characterized by a negative VOT and have greater voicing intensity. What is striking in Ruwund is that voiceless stops are also produced with voicing of weak intensity. These sounds should therefore be described as voiced stops. The presence and amount of aspiration during the delay between the first burst and the onset of voicing is the cue that contributes to these sound's identification as voiceless. Repp [1] observed that the increase in the amplitude of aspiration noise relative to the following periodic vocalic portion increases the salience of this cue and helps the probability to classify these consonants into the voiceless category. A perceptual test m confirms this hypothesis. Removing the part lying between the first burst and the following voiced part accounting for the beginning of the following vowel, show that listeners do not recognize these stops as voiceless anymore but as voiced. Ruwund reveal therefore new subtleties of voicing distinction and VOT. [1] B. Repp, "Relative amplitude of aspiration noise as a voicing cue for syllable-initial stop consonants," Lang. Speech, **23**, 173–189 (1979).

5aSCb7. San Juan Quiahije Chatino: A look at tone. Colette Feehan, Kelly Berkson, Malgorzata Cavar (Lingustics, Indiana Univ., 1021 E. Third St. Memorial Hall 322, IU Bloomington, IN 47405-7005, cmfeehan@umail.iu.edu), and Hilaria Cruz (Linguist, Univ. of Kentucky, Lexington, KY)

San Juan Quiahije (SJQ) Chatino is an under-documented indigenous language spoken in Oaxaca, Mexico, by some 3000 speakers. Like other members of the Eastern Chatino branch, SJQ has a complex tonal system—in particular, four tone levels and 11 lexical tones (14 when considering sandhi contexts). Excellent linguistic investigations of SJQ phonology (Cruz 2004, 2011) and discourse analysis (Cruz & Woodbury 2014, Cruz 2014) exist, but there is virtually no empirical phonetic work. We present exploratory and descriptive investigation of the tones of San Juan Quiahije Chatino. Data from one female native speaker largely align with the tonal description provided in Cruz (2011): our findings confirm the presence of a multitude of tones, and reveal that several lexical tones are merged in isolation. Furthermore, while depressor consonant effects—wherein tonal targets are lower after voiced obstruents than after voiceless obstruents—have traditionally been discussed primarily in the context of African languages (Bradshaw 1999), we discover a consistent lowering of tone by approximately 25 Hz after voiced consonants. The effect is present in both low and high tone contexts and perseverates throughout the vowel.

5aSCb8. An acoustic analysis of the vowels and stop consonants of Bashkir. Kelly H. Berkson, Matthew C. Carter, and Christopher M. Robbins (Linguist, Indiana Univ., 1021 E 3rd St., Memorial Hall 322 E, Bloomington, IN 47405, cartermc@umail.iu.edu)

Bashkir is a language of the Volga-Kipchak branch of the Turkic language family, spoken by approximately 1.2 million ethnic Bashkirs primarily in the autonomous Republic of Bashkortostan, Russia. Minimal research has been conducted on Bashkir in English, and what research has been conducted in either Russian or English focuses on the morphophonemics, syntax, and semantics of the language: acoustic investigation of Bashkir, meanwhile, is nearly nonexistent. This study is a preliminary examination of the phonetics of Bashkir. Using data from a female native speaker in her early 50s from Ufa, Bashkortostan, we present instrumental analysis of vowels in both pre-stressed and stressed positions, as well as voice onset time measures for oral stops. The vocalic data, in particular, are surprising in multiple ways: while they largely align with descriptions of the Bashkir vowel space provided by previous sources, with mid vowels that are reduced in most positions and that also have a large, variable range in the vowel space, they also suggest that the realization of one vowel phoneme may differ substantially from previous descriptions.

5aSCb9. The acoustics of strengthened glides in Kirundi. Alexei Kochetov (Linguist, Univ. of Toronto, 100 St. George St., Sidney Smith 4076, Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca)

Strengthening of post-consonantal glides /w/ and /j/ to obstruents or nasals is cross-linguistically uncommon, especially when the process is synchronically conditioned. This study investigates acoustic correlates of post-consonantal glide strengthening in Kirundi, an eastern Bantu language spoken in Burundi. The materials included words with /w/ or /j/ preceded by labial obstruent or nasal consonants /p, b, v, m/—the contexts that have been previously described to trigger the process. The morphological context was also varied, with sequences occurring at prefix-root or suffix-root boundaries, or within a root. Four female and four male native speakers produced the word list in a carrier sentence three times. The analysis involved phonetic classification of strengthened glides, as well as their duration and spectral measurements (frication/burst spectra and following vowel formants). The results showed that the post-consonantal /w/ was consistently realized as an oral or nasal stop agreeing with the preceding consonant in voicing and nasality, consistently with previous descriptive accounts. The realization of the post-consonantal /j/, however, was considerably more variable across and within speakers, ranging from a palatal stop or nasal to a weakly fricated/nasalized glide, and showing some sensitivity to morphological structure.

5aSCb10. Acoustic analysis of Punjabi stress and tone (Doabi dialect). Kiranpreet Nara (Linguist, Univ. of Toronto, 100 St. George St., Toronto, ON M5S 3G3, Canada, kiranpreet.nara@mail.utoronto.ca)

An acoustic experiment was conducted to study the stress and tone systems of Punjabi, an under-documented Indo-Aryan language. Tone and stress are linked because tone associates with the stressed syllable (Bailey, 1914; Wells and Roach, 1980; Baart, 2003). The experiment was used determine the acoustic cues of stress and the tonal contours of the three Punjabi tones: default, rising, and falling. Five native speakers read a list of 85 words five times. Measurements of duration, intensity, f0 were made in PRAATand analyzed in SPSS. The Mixed Models analysis of normalized intensity and duration revealed that the acoustic cue of stress is the duration of the rhyme. A similar finding for Hindi, a closely related language, is reported in Nair *et al.* (2001). As for tone, the default tone has the smallest f0 range and the falling tone has the largest f0 range. Falling tone is realized entirely on the stressed syllable whereas for the rising tone, the phenomenon of peak delay is observed unless tone occurs on a word-final syllable. Peak delay is also observed in Mandarin (Xu, 2001). This work offers an in-depth understanding of the phonological aspects of the stress and tone systems of Punjabi.

5aSCb11. Manner-specific tongue shape differences in the production of Kannada coronal consonants. Alexei Kochetov (Linguist, Univ. of Toronto, 100 St. George St., Sidney Smith 4076, Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca) and N. Sreedevi (Clinical Services, All India Inst. of Speech and Hearing, Mysore, Karnataka, India)

The production of consonants of the same place but different manner of articulation can involve certain adjustments in the posture of the tongue shape. This can be due to requirements for specific gestures (e.g., lowering the tongue sides for laterals) or constraints on coordination of different gestures (e.g., the tongue-palate constriction and the velum lowering for nasals). This study used ultrasound imaging to examine sagittal tongue shape differences in the production of Kannada (Dravidian) laterals, nasals, and stops of two places of articulation—alveolar/dental and retroflex. Words with these consonants (as geminates) were produced multiple times by five female and five male native speakers of Kannada. The analysis of tongue shapes revealed a lower tongue body/blade for laterals than stops, but only in retroflexes. The opposite was observed for /l/ vs. /t/, likely reflecting the alveolar vs. dental constriction differences. The nasals were produced with a significantly more advanced tongue body than the corresponding laterals and stops. The tongue fronting for nasals can serve to accommodate the velum lowering as part of these consonants' gestural coordination. The magnitude of this effect is further modulated by the consonant's degree of articulatory resistance, which is greater for retroflexes than alveolars/dentals.

5aSCb12. A sociophonetic account of morphophonemic variation in Palestinian Arabic. William M. Cotter (Linguist & Anthropology, Univ. of Arizona, 1009 East South Campus Dr., Tucson, AZ 85721, williamcotter@email.arizona.edu)

This study presents findings from sociolinguistic fieldwork on Palestinian Arabic in the Gaza Strip and Jordan. The sample includes 30 speakers representing three age groups, both genders, and three Palestinian communities: indigenous Gaza City residents, refugees from Jaffa who live in Gaza City, and Gaza City refugees living in Jordan. Linear mixed effects analyses are presented on the vowel raising of the Arabic feminine ending /ah/. The traditional dialect of Gaza realizes this morpheme consistently as [a] (Bergsträsser 1915), with all other Levantine city dialects raising the feminine ending to [e] or [i] except after back consonants (Al-Wer 2007). Similarly, the traditional Jordanian dialect in the area in which Gaza City refugees live also raises this vowel to [e] (Herin 2014). Results indicate robust sociophonetic variation in the realization of this vowel across these communities. Jaffa refugees in Gaza, whose traditional dialect realizes this vowel as [e], are found to be lowering their realization of this vowel with each successive generation, with younger speakers showing a phonetic realization similar to young indigenous Gazans. Simultaneously, Gaza refugees in Jordan show higher phonetic realizations across generations, indicating convergence toward the local realization of this vowel in Jordanian Arabic.

5aSCb13. A computational classification of Thai lexical tones. Jamison E. Cooper-Leavitt (Laboratoire de Phonétique et Phonologie - Sorbonne Nouvelle, Paris 3, 19 rue des Bernardins, Paris 75005, France, jecooper@ucalgary.ca)

The leveling of contour tones in Thai, uttered in a continuous context, has served as a natural point of difficulty for tone recognition experiments. Two tone recognition experiments presented here both include five lexical Thai tones (high, mid, low, rising, and falling) as abstract Bayesian models incorporated into a multi-model Hidden Markov Model. The HMM was developed using Thai natural language utterances to test its performance in correctly identifying Thai lexical tone categories. All utterances used for testing and training were produced in a laboratory setting. Utterances for the first experiment were produced in a citation context, and utterances for the second experiment were produced in a continuous context. The results of the two experiments were compared to test if the context had a significant effect on correctly identifying tone category. Findings showed the context of the utterance had a significant effect on the HMM's ability to correctly identify tone category, $F(1,48) = 5.82$, $p = 0.020$. The identification of the correct tone category for citation utterances showed a significant increase in performance than for the correct identification of tone category for continuous utterances, $t(48) = 5.38$, $p < 0.001$.

5aSCb14. Long-term average spectrum of te reo Māori. Donal Sinex (New Zealand Inst. for Lang., Brain, and Behaviour, Univ. of Canterbury, Utah State Univ., Logan, Utah 84322, don.sinex@usu.edu), Margaret Maclagan, and Jeanette King (New Zealand Inst. for Lang., Brain, and Behaviour, Univ. of Canterbury, Christchurch, New Zealand)

Te reo Māori, the language of the indigenous people of New Zealand, lacks some consonants common in English, notably the fricative /s/ and voiceless plosives. However, Māori pronunciation has adapted over time and exposure to English. Thus, there is reason to expect that the spectrum of spoken Māori differs from that of New Zealand English, but that the magnitude of the difference may be decreasing. We measured the long-term average spectrum (LTAS) of the speech of Māori-English bilinguals, using recordings from a longitudinal study of Māori speech. The oldest talkers in the database were born in the late 19th century. Female and male, talkers, and both native and non-native speakers of Māori were included. The LTAS was determined for each talker and each language. For individual talkers, the average spectrum had consistently higher amplitude for English than Māori, for frequencies above approximately 2 kHz. Across talkers, the mean difference at 8 kHz was approximately 8 dB, for all but one group. The exception was younger female Māori L2 talkers, for whom the difference was only 3 dB. These observations are consistent with the consonant inventory of te reo Māori, and they also provide potentially useful information about ongoing changes in the language.

5aSCb15. Production and perception of the advanced tongue root vowel system in Ethiopian Komo. Paul Olejarczuk, Manuel A. Otero, and Melissa M. Baese-Berk (Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, paulo@uoregon.edu)

Komo [xom] is an endangered and under-documented language spoken along the Ethio-Sudanese border. This paper presents the results of the first phonetic investigation of the Komo vowel system and reports on a related perception experiment carried out in the field. Our first aim is to provide an acoustic description of the Advanced Tongue Root (ATR) feature in Komo, which is contrastive in the high vowels and allophonic in the non-high vowels. To this end, we present acoustic measurements of 2,688 vowel tokens produced by 16 speakers. Our second aim is to examine the influence of Komo's typologically unique vowel harmony system on listeners' perception of the [ATR] feature. Komo ATR harmony displays two competing processes triggered by the high vowels (Otero, 2015): [+ATR] spreads leftward to non-high vowels (e.g., /CaCi/ → [CəCi]), while [-ATR] spreads rightward to high vowels (e.g., /CɪCi/ → [CɪCɪ]). Sixteen Komo listeners and 16 native English-speaking controls performed an AX task with disyllabic (pseudo)-words featuring context vowels and [±ATR] vowel continua. Patterns in the results suggest that (a) Komo listeners (but not controls) were influenced by the [ATR] value of the context, and (b) targets of [+ATR] harmony were processed differently from targets of [-ATR] harmony.

5aSCb16. Preliminary acoustic descriptions of the pharyngeals and sosterior plosives of Northern Haida. Corey Telfer and Jordan Lachler (Linguist, Univ. of AB, 2-40 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, telfer@ualberta.ca)

Haida is a highly endangered language spoken on the archipelago of Haida Gwaii, off the coast of British Columbia, as well as in communities in Southeast Alaska. Like many languages of the Northwest Coast, Haida plosives include three manners of articulation: voiceless aspirated, voiceless unaspirated, and ejective. Analysis of recordings indicates that the Voice Onset Time of the posterior aspirated and ejective stops is nearly identical. It appears that these speech sounds differ primarily in their acoustic intensities, and a new measure called Burst Intensity Slope is proposed to quantify this difference. In addition, the Northern dialect has been described as including pharyngeal speech sounds (e.g., Krauss 1979, Enrico 1991); however, only one small acoustic study has been conducted to verify this (Bessell 1993). Using recordings of a small number of speakers, this paper aims to document the different types of pharyngeals using acoustic measurements. Of special interest is the pharyngeal plosive of Massett Haida, which often includes what appears to be concomitant aryepiglottal trilling ("growl voice"). This will be investigated by comparing the number of zero-count crossings with those of other types of plosives and vowels produced by the same speakers.

5aSCb17. Prevoicing differences in Southern English: Gender and ethnicity effects. Wendy Herd, Devan Torrence (MS State Univ., 100 Howell Hall, PO Box E, MS State, MS 39762, wherd@english.msstate.edu), and Joy Cariño (MS School of Math and Sci., Columbus, MS)

Differences in the way VOT is used across languages to maintain stop voicing contrasts have been well-documented, but less research has been focused on VOT variation within voicing categories. For example, native English speakers are generally reported to produce word-initial voiced stops with short positive VOTs, but within category gender and ethnicity differences have been reported in one preliminary study, with male speakers prevoicing stops more than female speakers and with African American speakers prevoicing stops more than Caucasian American speakers (Ryalls, Zipprer, and Baldauff, 1997). For the current study, native speakers of English from Mississippi were recorded reading three repetitions of a pseudo-randomized list of words designed to investigate the effects of gender and ethnicity on the prevoicing of word-initial voiced stops. Participants self-identified their gender and ethnicity in a language background survey completed after recordings. Significant ethnicity, but not gender, differences were found. Strikingly, African American speakers produced voiced stops with prevoicing approximately 90% of the time, while just 35% of the voiced stops produced by European American speakers were prevoiced. These findings strongly suggest that dialectal differences play a role in within category variation in the VOTs of word-initial voiced stops.

5aSCb18. Weight-sensitive stress and acoustic correlates of disyllabic words in Marathi. Esther S. Le Grezause (Linguist, Univ. of Washington, 4300 Woodland Park Ave. N APT 203, Seattle, WA 98103, elg1@uw.edu)

Little of the literature on Marathi phonology addresses stress, and none of it systematically. The tentative accounts that do exist are contradictory in their descriptions (Pandharipande 1997, Dhongde and Wali 2009). Moreover, there is no research on the acoustic correlates of Marathi stress. To address this gap, this study investigates word-level stress and the acoustic correlates of stress in disyllabic words in Modern Standard Marathi. Since weight criteria vary across languages (Hayes 1995, Gordon 2004), the main goals are to determine what constitutes heavy syllables and whether or not long vowels behave like heavy vowels with two moras ($\mu\mu$). Three main hypotheses are tested: (H1) stress is weight-sensitive, (H2) Marathi follows the Latin weight system (Gordon 2002) with CVX heavy, and (H3) when syllables have the same weight the left-most syllable bears stress. The study tests these hypotheses using auditory and spectrographic analyses of recordings from two native Marathi speakers. Results confirm all three hypotheses: H1: Marathi has weight-sensitive stress, H2: it follows the Latin weight distinction and, H3: the first syllable is stressed when syllable weight is equal. Preliminary acoustic results indicate that pitch, F1, and F2 are slightly higher in stressed syllables than in unstressed syllables.

5aSCb19. A preliminary analysis of stop codas in Kandze Khams Tibetan. Vitor Leongue (Linguist, Indiana Univ., 1021 East 3rd St. – Memorial Hall 322 E, Bloomington, IN 47405, vleongue@indiana.edu)

Research on Tibetan linguistics has been scarce. There are very few systematic accounts of the sounds in the language, and acoustic-phonetic analyses, in particular, are very difficult to come by. Furthermore, much of the existing literature has focused on the Lhasa dialect; many other varieties are not as adequately described. The current study investigates the phonetic realization of coda consonants in the under-documented Kandze dialect of Khams Tibetan. Historically, Tibetan allows up to two consonants in coda position, as reflected in the written language and in some western varieties. In most modern Tibetan dialects, however, many of the codas have been lost or simplified in some way. Preliminary data from one male native speaker indicate that codas that were historically voiced oral stops are realized in Kandze as glottal stops word-finally. When occurring word-internally, these codas tend to undergo deletion, which may then trigger lengthening of the preceding vowel. In both final and internal positions, some of the codas may also trigger vowel fronting. Acoustic data are presented to illustrate the alternations exhibited by the codas and the vowels that precede them.

5aSCb20. A preliminary account of the Thangal sound system. Patricia McDonough, Erin Arnold, and Kelly Berkson (Linguist, Indiana Univ., 1021 East 3rd St. – Memorial Hall 322 E, Bloomington, IN 47405, pmcdonou@indiana.edu)

This work presents instrumental acoustic analysis of Thangal, a severely under-resourced Tibeto-Burman language spoken primarily in the Senapati District of Manipur by around 2500 people. While some research has been conducted in recent years on related languages such as Rongmei, and several Thangal wordlists compiled mainly from data collected in the 1800s also exist, linguistic investigation of Thangal in the past century has been limited at best, and empirical phonetic investigation is nonexistent. We use data from wordlists produced by native Thangal speakers to compile a preliminary phonetic inventory of the language and map the Thangal vowel space. Initial notable observations include the existence of prenasalized plosives and both aspirated and unaspirated consonants. This work was undertaken at the request of the community, whose ultimate hope is to develop a practical orthography in service to developing written materials in Thangal.

5aSCb21. Affricate contrasts in Déline Slavey. Maida Percival (Linguist, Univ. of Toronto, 100 St. George St., 4th Fl., Toronto, ON M5S 3G3, Canada, maida.percival@mail.utoronto.ca)

Dene or North Slavey is a Dene (Athabaskan) language with nine affricates: /ts, ts^{h}, ts’, tʃ, tʃ^{h}, tʃ’, tɬ, tɬ^{h}, and tɬ’/. This paper provides a first investigation into the acoustics of these sounds using ~2000 tokens from eight speakers from Déline, NT, Canada. Within the lateral series, [tɬ] was found to be realized in a number of ways including [tɬ], [tɬl], [tl], and [tɬɮ], with phonetic voicing to varying degrees in the release. In contrast, [tɬ^{h}] was generally produced with a brief period of aspiration following the frication portion of the affricate, while ejectives were often followed by a brief period of silence between the frication and the vowel onset. COG measurements of [tɬ] (from the voiceless portion of the release) were lower than the alveolar and post-alveolar affricates, averaging 3500 Hz (s.d. = 2175 Hz). However, COG variation across and within speakers and words suggests varying place of articulation. Place of articulation may be less crucial to the identification of these sounds than manner cues such as the lateral component. Further work is underway to investigate differences in the distribution of energy in the frication and F0 perturbations between the different phonation types, and to investigate the alveolar and post-alveolar affricates in more depth.

5aSCb22. Acoustic characteristics of vowels in two Totonacan languages. Rebekka Puderbaugh (Dept. of Linguist, Univ. of AB, 2-40 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, puderbau@ualberta.ca)

This study examines the acoustics of vowels from two Totonacan languages, Upper Necaxa Totonac (UNT) and Huehuetla Tepehua (HT). Both languages have five-vowel systems consisting of the qualities /aeiou/ as well as phonemic quantity distinctions (short, long), and lexical stress. In addition, UNT makes use of contrastive phonation on vowels, while HT vowels may be produced with allophonically non-modal phonation when adjacent to glottalic segments and glottal stops. Data from four speakers (two female and two male) of each language are reported. Acoustic measures are based on multiple repetitions of each vowel in a variety of lexical items, elicited within a frame sentence. Measurements are taken from stressed vowels surrounded by obstruents wherever possible. A variety of analyses are undertaken. Traditional visualizations of the vowel spaces of male and female speakers using normalized F1-F2 plots are combined with comparisons of dynamic formant trajectories across the time course of vowel production. Vowel duration is compared across short and long vowel categories of all five qualities in both languages. The relationship of fundamental frequency to vowel identity and phonation type is investigated.

5aSCb23. Arabic as an under-documented language: Distinctions between neighboring Arabic dialects. Judith K. Rosenhouse (Linguist, SWANTECH, 9 Kidron St., Haifa 3446310, Israel, judith@swantech.co.il)

This paper focuses on Arabic dialects from Israel and Jordan, which belong to the Levant Arabic dialects group. Levant dialects, as part of Arabic, are rather well studied. However, studies still usually consider them following the language situation in the area before the end of World War I

(1918). Then, the Ottoman Empire collapsed and the area of (current) Israel and Jordan came to be under the British mandate rule. Later, each of them won independence as separate states. These processes widened inter-dialect linguistic differences, but modern communication media and devices, and increased literacy, have drawn various dialects closer to one another. These processes have added new linguistic features, which have in various cases completely integrated in the borrowing dialects. As a result, these features cannot be distinctive dialect markers any more. Such developments are mostly under-documented. Phonetic features of vowels and consonants in some Israeli and Jordanian Arabic dialects will demonstrate this phenomenon of under-documentation. In this context, we also discuss the forensic linguistics field of dialect identification and verification of asylum seekers' mother tongue. Our conclusions suggest that these undocumented changes require new investigations.

5aSCb24. The status of voiceless nasals in Miyako Ryukyuan. Catherine Ford, Benjamin V. Tucker, and Tsuyoshi Ono (Univ. of AB, 2-40 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, bvtucker@ualberta.ca)

The current study investigates voiceless nasals in Miyako, a language spoken on remote Japanese islands near Taiwan, focusing on the dialect Ikema. Voiceless nasals are rare cross-linguistically. Hayashi (2013) suggested that there is phonemic contrast between voiced /n, m/ and voiceless /n̥, m̥/. Pilot data from one speaker both confirmed and refuted Hayashi's claims, analysis indicating that these voiceless nasals occur both as voiceless and breathy voiced. Thus, the glottal state of this phoneme may depend on the phonetic environment. The current study further explores these findings based on data collected from four male and two female speakers. Participants were asked to produce target and minimal pair words in isolation and within novel sentences. Voiceless and voiced nasal segments were compared and analyzed for duration, cepstral peak prominence, and amplitude measures of H1 and H2 to determine whether breathy voicing state might differentiate target phonemes from /n/ and /m/. The results of this analysis are used to investigate the nature of this phonemic contrast and how Ikema is situated cross-linguistically in its use of voicing to distinguish nasal phonemes.

5aSCb25. Pitch specification of minor syllables in Sgaw Karen. Luke West (Linguist, Univ. of California, Los Angeles, 785 Weyburn Terrace, Apt. 421, Los Angeles, CA 90024, lukewest@g.ucla.edu)

Sgaw Karen is a Tibeto-Burman language with both lexical tone and a major-minor syllable divide. Unlike major syllables, minor syllables are structurally reduced and do not bear overt phonological tone. This acoustic study investigates the fundamental frequency of these minor syllables, summarizing surface pitch patterns and providing evidence for a phonetic target. The first acoustic look at minor syllables in Karenic languages, a phonetically understudied language family, 58 unique minor syllable sequences of each possible length (~500 tokens) are elicited from one native speaker from the Karen State, Myanmar (Burma). Analysis of minor syllable sequences adjacent to lexical tones shows (1) a low pitch target around 185 Hz to which minor syllables asymptotically approximate when given sufficient time, and (2) this target is not explicable by contextual interpolation. Results address the general problem of tonal underspecification and interpret findings in the context of a minor syllable typology. Acoustic surface patterns of the Sgaw Karen minor syllable point to phonetic pitch targets of gradient strengths. Findings provide a foundation for detailed tonal/intonational analysis of Sgaw Karen, and demonstrates a framework for acoustic documentation of underspecified tonal units in understudied languages of Southeast Asian in general.

5aSCb26. On the presence of voiceless nasalization in apparently effaced Somali Chizigula prenasalized stops. Michal Temkin Martinez and Haley K. Boone (English, Boise State Univ., 1910 University Dr., Boise, ID 83725-1525, haleyboone@u.boisestate.edu)

Somali Chizigula (G311; xma; also Mushungulu) is an endangered language spoken in Somalia and by Somali-Bantu refugees abroad. We report on an acoustic and aerodynamic study of N + stop sequences in Somali Chizigula. Our findings illustrate that while the only acoustic cue for the nasal in voiceless prenasalized stops is aspiration of the oral stop, appearing to have undergone total effacement, aerodynamic data show robust nasal airflow prior to the oral stop. Although nasal devoicing is not rare in Bantu languages (e.g., Bura and Bondei; Maddieson and Ladefoged 1993), the lack of acoustic cues during the nasal is. The current state of the devoiced nasal in Somali Chizigula could lead to a reanalysis of effacement as [nt → nt^{h} → (**n̥t$^{\mathbf{h}}$**)→ t^{h}]. In addition to providing acoustic and aerodynamic data, we propose an articulatory phonology (Browman and Goldstein 1993) account for the voiceless prenasalized stop as it compares to other nasal + stop sequences in the language.

5aSCb27. Word length is (in part) predicted by phoneme inventory size and syllable structure. Ian Maddieson (Dept. of Linguist, UNM, Univ. of New Mexico, MSC03-2130, Albuquerque, NM 87131-0001, ianm@berkeley.edu)

Based on anecdotal data from a few languages, it has been proposed that typical word length in a language is correlated with the size of the phoneme inventory: more phonemes predicts shorter words. This hypothesis has now been examined in some detail using a substantial sample of languages. A standardized text widely employed for phonetic illustrations is used, namely, the fable of "The North Wind and the Sun." Illustrations published by the IPA and others have been collected for over 100 languages. These illustrations are arguably well-matched in style and vocabulary and reflect natural inflected forms embedded in text. Average word-length in these texts is calculated and correlated with the size of the language's consonant and vowel inventories, as well as an index reflecting the permitted complexity of syllable structure. Vowel inventory is treated in two ways; basic vowels (just the distinctions on the basic height, backness and rounding parameters), and total vowels (including other distinctions such as nasalization, length, and voice quality). Mean word length is indeed negatively correlated with size of consonant and vowel inventories, most strongly with the number of basic vowels. More complex syllable structures also correlate with a shorter mean word length.

Session 5aSCc

Speech Communication: Speech Production (Poster Session)

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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.

5aSCc1. Simulations of three-dimensional, self-oscillating vocal fold replicas with liquid-filled cavities. Eduardo J. Alvarez and Scott L. Thomson (Mech. Eng., Brigham Young Univ. - Idaho, BYU-Idaho, Rexburg, ID 83460, Alv14011@byui.edu)

Synthetic, self-oscillating vocal fold replicas are used in bioreactors to study the response of injected cells to phono-mimetic vibrations. In this type of replica, cells are contained within a cylindrically shaped cavity that runs anterioposteriorly and that is located near the replica surface. During vocal fold replica flow-induced vibration, the cavity deforms, thereby subjecting the cells to a periodically varying stress field. The characteristics of this stress field are unknown, and experimental determination is not presently feasible. Therefore, in this research, numerical simulations of these self-oscillating, cell-containing synthetic replicas are studied. A three-dimensional vocal fold model containing a liquid-filled cavity, developed using the commercial software package ADINA, is described. The effects of glottal airflow are modeled using the mechanical energy equation to simulate the pressure distribution along the replica airway surface. The liquid-filled cavity is modeled through a fully coupled fluid-structure interaction solver, with the liquid region governed by the viscous, unsteady, three-dimensional Navier-Stokes equations. Estimates of the stress field within the cavity as well as the predicted influence of the cavity on replica vibration will be reported.

5aSCc2. What the 'L?: An ultrasound study of the acoustic and articulatory characteristics of laterals in Brazilian Portuguese. Sherman D. Charles (Linguist, Indiana Univ., 1610 S Dorchester Dr. Apt. 49, Bloomington, IN 47401, sdcharle@indiana.edu)

Recent technological and computational advances have enabled researchers to investigate vocal tract articulation and acoustics with an ever-increasing degree of nuance and specificity. This is particularly enlightening in the case of articulations which are subject to tremendous cross-linguistic and interspeaker variability, like laterals (Ladefoged and Maddieson 1996). The current study uses simultaneous 3D ultrasonography and audio recordings to investigate vocal tract configurations and their acoustic consequences, and to provide a comprehensive descriptive analysis of alveolar and palatal lateral speech sounds produced by one Brazilian Portuguese speaker. The results confirm previous accounts of multiple articulatory constrictions and low F1 and F2 frequencies for alveolar laterals (Barbarena *et al*. 2014; Johnson 2012; Ladefoged 2003; Ladefoged and Maddieson 1996). Novel contributions include (1) identification of a dynamic tongue gesture that, based on a Perturbation Theory analysis (Stevens 1996), appears to directly reflect formant transitions from a preceding vowel to the alveolar lateral consonant, and (2) three-dimensional articulatory descriptions of both alveolar and palatal laterals in Brazilian Portuguese.

5aSCc3. Articulator movement contributions to formant trajectories in diphthongs. Christopher Dromey and Katherine McKell (Commun. Disord., Brigham Young Univ., 133 TLRB, BYU, Provo, UT 84602, dromey@byu.edu)

Our purpose was to learn about the relative contributions of lip aperture and protrusion as well as tongue height and advancement to the trajectories of F1 and F2 during diphthongs produced by 20 speakers. Audio and kinematic signals were recorded with an NDI Wave system. Audio segments were analyzed with PRAAT to extract the F1 and F2 histories during the diphthongs. These were time-aligned with the kinematic records in MATLAB. Because time-series data violate the assumptions of traditional correlation and regression analyses, novel methods were developed to represent the relative contributions of individual articulator movements to F1 and F2 changes. All movement and formant tracks during diphthong transitions were converted to z-scores and overlaid on a common time scale. Absolute difference scores between kinematic (predictor) variables and acoustic (dependent) variables were summed along each plot to reflect the strength of the contribution of each movement to the acoustic outcome. The lowest difference sums reflected the closest alignment between a movement and a formant track. Results revealed differences in the relative contributions of the tongue and lips for each diphthong, suggesting that formant patterns during diphthongs may be difficult to interpret in a straightforward way in terms of articulator movement.

5aSCc4. Rhotic articulation by first graders: A real-time three-dimensional ultrasound study. Olivia Foley, Sarah McNeil, Katherine Schlimm, Amy Piper, and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, oliviarfoley@gmail.com)

The rhotic /r/ in American English is a complex sound that is produced differently by different speakers and in different phonetic contexts, but there is currently little knowledge about how children articulate this sound, which is typically acquired late in development. In this study, the Goldman-Fristoe Test of Articulation (GFTA-3) was administered to typically developing first graders (6 and 7 years old) as part of a 5-year longitudinal study. The GFTA-3 contains words in which /r/ occurs in a variety of syllable positions and phonetic contexts. Using 3D ultrasound imaging and palate impressions, it is possible to view the tongue shape in relation to the speaker's palate. Since first graders are younger than the age at which /r/ is usually completely acquired, this presents an opportunity to see how /r/ is articulated by children who may not yet have mastered it. Initial results from this study will be presented, with the aim of empirically characterizing tongue shape and motion in /r/ production by first graders.

5aSCc5. Velopharyngeal status of vowels produced with and without hard glottal attack in children with repaired cleft palate. Marziye Eshghi (Speech, Lang. and Hearing Sci., Craniofacial Ctr., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu) and David J. Zajac (Dental Ecology, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

The objective of this study was to describe velopharyngeal (VP) closure patterns of isolated vowels produced by infants with repaired cleft palate (CP). Nasal ram pressure (NRP) and audio signals were obtained from six infants with repaired CP (four males and two females). Two infants were 12 months of age and four were 18 months of age. All infants were from American-English speaking families without known syndromes. A total number of 95 isolated vowels were analyzed perceptually and spectrographically. Fourteen vowels were identified as produced with hard glottal attack. Three

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infants produced all vowels with at least 96% complete VP closure. Three other infants produced all vowels with 59% complete VP closure and 39% partial VP closure. In addition, one of the infants produced an isolated vowel with a completely open velopharynx. Vowels with hard glottal attack, however, were produced with 100% complete VP closure across all subjects, regardless of VP status for vowels without hard glottal attack. It is hypothesized that early use of hard glottal attack by children with repaired CP might contribute to the often reported high prevalence of voice disorders and may even be associated with the development of glottal stop articulation. [Research reported in this publication was supported by the National Institute of Dental & Craniofacial Research of the National Institutes of Health under Award Number 1R01DE022566-01A1.]

5aSCc6. Event-related cortical potentials occurring prior to speech initiation. Silas Smith, Maira Ambris, Serena Haller, and Al Yonovitz (Dept. of CSD, Univ. of Montana, Missoula, MT 59812, silas.smith@umontana.edu)

Real time brain electrical potentials were obtained in subjects prior to the initiation of speech in a potential clinical paradigm using a single vertex electrode. The purpose of this research was to establish a real-time event related brain electrical potential system. Research in this area in the past has used a great number of facial and lip EMG electrodes. The marking point for determining the pre-event time epoch has been an EMG source. The data are typically acquired off-line and later averaged. This research uses a vocal signal as the marking point and displays in real time the event-related potential. The sample rate (25,600 samples/s) permitted an analysis of both slow negative waves and faster neurogenic signals. Results indicated reliable waveform morphology within and between subjects.

5aSCc7. Nasal coarticulation in normal infants: A physiological analysis. Marziye Eshghi (Speech, Lang. and Hearing Sci., Craniofacial Ctr., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu) and David J. Zajac (Dental Ecology, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

There are conflicting views regarding coarticulation in speech of children. Kent (1983) suggested that children's speech is more "segmental" while Nittrouer *et al.* (1989) proposed that speech of young children has "syllabic organization" with syllables being units of speech production. According to Nittrouer *et al.* (1989), children may demonstrate more coarticulation than adults. The purpose of this study was to examine nasal coarticulation in the speech of infants over time. Nasal ram pressure (NRP) and audio signals were obtained from ten typically developing infants (five males and five females) at 12, 14, and 18 months of age. Six subjects produced 39 nasal syllables (either NV, VN, NVN, or VNV) at 12 months of age. Six subjects produced 62 nasal syllables at 14 months of age and eight subjects produced 95 nasal syllables at 18 months of age. Results revealed positive NRP at the midpoint of only 20% of vowels either preceding or following nasal consonants at 12 months of age. Positive NRP, however, was present during 62% and 77% of vowels produced at 14 and 18 months of age, respectively. Findings provide evidence for the development of nasal coarticulation over time from a more "segmental" to a "syllabic" level. {Research reported in this publication was supported by the National Institute of Dental & Craniofacial Research of the National Institutes of Health under Award Number 1R01DE022566-01A1.]

5aSCc8. Muscular structure of the human tongue from magnetic resonance image volumes. Maureen Stone (Univ. of Maryland Dental School, 650 W. Baltimore St., Rm. 8207, Baltimore, MD 21201, mstone@umaryland.edu), Jonghye Woo (Massachusetts General Hospital, Boston, MA), Junghoon Lee (Johns Hopkins School of Medicine, Baltimore, MD), Tera Poole, Amy Seagraves, Michael Chung, Eric Kim (Univ. of Maryland Dental School, Baltimore, MD), Emi Z. Murano (Hospital das Clinicas da Faculdade de Medicina, Sao Paulo, Brazil), Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD), and Silvia S. Blemker (BME, Univ. of Virginia, Charlottesville, VA)

The human tongue has a complex architecture, consistent with its complex roles in eating, speaking, and breathing. Tongue muscle architecture has been depicted in drawings and photographs, but not quantified volumetrically. This study aims to fill that gap by measuring the muscle architecture of the tongue for 14 people captured in high-resolution 3D MRI volumes. The results show the structure, relationships, and variability among the muscles, as well as the effects of age, gender, and weight on muscle volume. Since the tongue consists of partially interdigitated muscles, we consider the muscle volumes in two ways. The functional muscle volume encompasses the region of the tongue served by the muscle. The structural volume halves the volume of the muscle in regions where it interdigitates with other muscles. Results show similarity of scaling across subjects, and speculates on functional effects of the anatomical structure.

5aSCc9. An electromagnetic articulometer study of tongue and lip troughs. Christine H. Shadle (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), Hosung Nam (Haskins Labs., Seoul, South Korea), Argyro Katsika, Mark Tiede, and D H Whalen (Haskins Labs., New Haven, CT)

Troughs, which are a discontinuity in anticipatory coarticulation (Perkell 1968), have been shown to occur in the tongue body in /ipi/ and lips in /usu/ . An electromagnetic articulometer (EMA) study of a single subject reported previously showed that intraoral pressure during the consonant could not be solely responsible for the downwards tongue body movement observed during the labial consonants (/p, b, m, f, v/); results supported the hypothesis of a secondary tongue gesture during the consonant. Lip troughs during /s/ in rounded contexts were more ambiguous, possibly because of inconsistencies in corpus design. Six new subjects have been recorded using a WAVE EMA system and a revised corpus. Tongue trough magnitudes decreased in the order /f, v/ $>$ /p, b/ $>$ /m/, serving to disprove the aerodynamic hypothesis; they were deeper for long than short consonants (e.g., [iffi] $>$ [ifi]), and converged to the same point in asymmetric vowel contexts. The same type of convergence during the consonant was observed for lip aperture during $V_1C(C)V_2$ combinations, where $C = \{/s, ʃ/\}$. These recent results thus support both tongue and lip troughs being caused by secondary gestures of consonants. [Work supported by NIH NIDCD-DC-02717.]

5aSCc10. Underlying mechanism for nasal rustle noise during speech in children with cleft palate. Hedieh Hashemi Hosseinabad, Suzanne Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267-0379, hashemhh@mail.uc.edu), and Ann W. Kummer (Div. of Speech-Lang. Pathol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH)

Nasal rustle (AKA Nasal turbulence) occurs as an obligatory nasal air emission in children with cleft palate. It usually occurs as a result of air passing through a partially small velopharyngeal port. As the air moves through this small opening, the air pressure increases. When this pressurized air is released on the nasal side of the opening, it causes nasal secretions on top of the valve to bubble. The hypotheses is that nasal rustle is due to audible bubbling of secretions; so when secretions are removed, the extra noise disappears. In this study, we applied signal to noise ratio (SNR) before suctioning the secretions to explore the relationship between presence of noise and bubbling. Ten children with nasal rustle were recorded using the microphones of the Nasometer II during production of high pressure sentences pre-post suctioning. Results will be discussed. It is expected that the findings reveal that using SNR can be used clinically as an acoustic parameter to detect the effect of bubbling in nasal rustle.

5aSCc11. Complete tonal neutralization in Taiwan southern Min. Mao-Hsu Chen (Linguist, Univ. of Pennsylvania, 4200 Spruce St., Apt. 310, Philadelphia, PA 19104, chenmao@sas.upenn.edu)

This study reported the result of a production experiment on the tonal neutralization of context unchecked 21 tone and checked 21 tone with a glottal stop coda in Taiwan Southern Min, which are said to be realized the same as a high falling sandhi tone 51 on surface when occurring in context positions. Speakers of two age groups, younger and older, were recruited for the production experiments to examine whether the neutralization is complete. The comparison between the f0 contours was conducted with smoothing spline analysis of variance, and it showed a case of complete neutralization for both age groups, where the two f0 contours were simply

overlapped. As reported by linear mixed effects modeling, the durations of the target syllables with underlying checked and unchecked 21 tones were not significantly different from each other.

5aSCc12. Interaction between aspiration and tonal pitch in Weihai Chinese. Jie Deng and Sean A. Fulop (Linguist, California State Univ. Fresno, 5245 N Backer Ave., PB92, Fresno, CA 93740-8001, sfulop@csufresno.edu)

This paper considers the questions of whether, how, and by what mechanism the aspiration of voiceless stops in the syllable onset affects the pitch (F0) of lexical tone in a tone language. The literature on this topic is at odds with itself. Lai *et al.* ["The raising effect of aspirated prevocalic consonants on F0 in Taiwanese," *Proceedings of the 2nd International Conference on East Asian Linguistics* (2009)] review the history of conflicting results, with some studies reporting F0 lowering after aspirated stops, and some reporting F0 raising. Here is provided another battery of findings from another tonal dialect, Weihai Chinese. Using data from seven monodialectal female speakers, no effect of aspiration is found on tonal F0 following bilabial or alveolar stops, but a strong significant lowering effect was observed following the velar aspirated stops. A second issue addressed here has had almost nothing published about it, viz., what is the converse effect of the lexical tone pitch upon the aspiration time (VOT) of the onset stops. It was found that, in general, high-beginning tones correlated with shorter voice onset times in all stop onsets.

5aSCc13. The role of f0 on acquisition of a phonological contrast in Korean stop system. Gayeon Son (Linguist, Univ. of Pennsylvania, 619 Williams Hall, 255 S 36th St., Philadelphia, PA 19104, gson@ling.upenn.edu)

A number of studies have investigated the role of Voice Onset Time (VOT) on acquisition of stop voicing contrast. Korean stop contrasts (lax, tense, and aspirated), however, cannot be differentiated only by VOT since they are all pulmonic egressive voiceless stops. For this three-way distinction, another acoustic parameter, f0, critically operates. The present study explores how f0 is perceptually acquired and phonetically operates for Korean stop contrast over age. In order to reveal the relationship between f0 developmental patterns and age, a quantitative acoustic model dealt with word-initial stop productions by 60 Korean young children aged 20 months to 47 months. The results showed that f0 perceptual acquisition patterns are closely related to children's articulatory distinction especially between lax and aspirated stops. In the case of aspirated stops, phonetic accuracy depends on the perceptual thresholds in f0, and the significant phonetic differentiation between lax and aspirated stops was found at age over 32 months. However, f0 of tense stops is not significantly distinguished from the other two stop categories in production. These findings suggest that acquisition of f0 plays a crucial role in the formation of phonemic categories for lax and aspirated stops and this process significantly affects articulatory distinction.

5aSCc14. The physiological underpinnings of vowel height and voice quality. Laura Panfili (Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195, lpanfili@uw.edu)

This study examines the distribution of creaky voice across vowel heights and discusses the physiological mechanisms that may influence this distribution and its methodological implications. The data for this study come from eight dyads from the ATAROS Corpus of audio-recorded conversations between Pacific Northwest English speakers (Freeman 2015). Stressed vowels in content words were tagged for phonation type based on auditory judgments. A chi square test of independence ($\alpha = 0.01$) found a significant relationship between vowel height and creak ($\chi^2(1, N = 2459) = 83.58$, $p < 0.001$), such that low vowels were more likely to be creaky than high vowels. This effect may be due to the same physiological mechanisms as Intrinsic Fundamental Frequency (IF0). Though the exact mechanism behind IF0 is unclear, various hypotheses have suggested that the tongue position required for high vowels pulls on the larynx, increasing tension, decreasing mass, and resulting in a higher F0 (Ladefoged 1964, Lehiste 1970). These laryngeal settings would also disfavor creaky voicing, perhaps explaining the distribution of creaky voice and vowel height. Future studies of phonation should consider vowel height in their methodologies.

5aSCc15. Representations of electromagnetic articulography data for tongue shaping and vocal tract configuration. Sungbok Lee (Elec. Eng., Univ. of Southern California, 3601 Watt Way, GFS-301, Los Angeles, CA 90089, sungbokl@usc.edu), Dani Byrd (Linguist, Univ. of Southern California, Los Angeles, CA), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Electromagnetic articulography (EMA) measurements of the movements of articulator flesh points have predominantly been used to investigate kinematics of individual points and/or coordinative phasing between them. Such limited use of EMA data is in fact wasteful of information. In this preliminary report we introduce two representations of EMA data: one is a representation of lingual configuration derived from three individual flesh points on the tongue, and the other is a representation of the vocal tract configuration information based on a lower-dimensional representation of Euclidean distances between EMA articulatory points. The motivation is to maximize the use of information in the original EMA data and measurements in a way that allows for the representation of tongue shaping and vocal tract configuration, thereby maximizing the use of information available from EMA and basic EMA measurements. Implications of such representations and their patterning will be discussed in the light of speech production dynamics as functions of speech rate and categorical emotions expressed in speech. [Work supported by NIH DC03172 and NSF IIS-1116076.]

5aSCc16. Measuring contact area in synthetic vocal fold replicas using electrical resistance. Kyle L. Syndergaard, Stephen Warner, Shelby Dushku, and Scott L. Thomson (Mech. Eng., Brigham Young University-Idaho, BYU-Idaho, Rexburg, ID 83460, kyle.syndergaard@gmail.com)

Synthetic vocal fold replicas are a useful tool for studying human voice production. These replicas are often studied using high-speed imaging, but additional analysis procedures are desired that are lower cost, less data intensive, and more conducive to long-term vibration monitoring. Consequently, this work explores the use of electrical resistance to measure vocal fold contact area across the glottis of a synthetic vocal fold replica. The concept is based on electroglottography (EGG), a clinical tool for measuring vocal fold contact area *in vivo*. When a small current is passed through the vocal folds, the resistance across the folds varies inversely with the contact area between the folds. This change in resistance thus makes it possible to estimate the degree of glottal closure during vocal fold vibration. For this research, a method of enabling silicone vocal fold replicas to conduct electricity without compromising the sensitive material viscoelastic properties has been developed. The concept will be demonstrated and relationships between contact area and resistance in both static and vibrating (self-oscillating) replicas will be presented.

5aSCc17. Prosodic strengthening at the edges of prosodic domains in sighted and blind speakers. Lucie Menard, Pamela Trudeau-Fisette, and Melinda Maysounave (Linguist, Universite du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

Recent studies have shown that when producing isolated vowels, congenitally blind speakers produce smaller displacements of the lips (visible articulators), compared with their sighted peers. To investigate the role of visual experience on articulatory gestures used to produce salient speech contrasts, the production of vowels at the edges of low-level prosodic domains and high-level prosodic domains (in which gestures are reported to be strengthened) was studied in adult speakers of Quebec French. Ten sighted and ten congenitally blind participants were recorded during the production of the vowels /i/, /y/, /u/, /a/ in initial and final positions of two prosodic domains: word and intonational phrase. Synchronous acoustic and articulatory data were recorded using the Carstens AG500 Electromagnetic Articulograph system. Formant measures as well as displacements of the lips and tongue were analyzed. The results revealed that speakers produced larger ranges of lip and tongue movements at the edges of higher prosodic domains than at the edges of lower ones. Formant spaces and durational data are presented. The use of visible (lip and jaw) and invisible (tongue) articulators to implement the prosodic structure is discussed

5aSCc18. Three-dimensional tongue shapes of /r/ production in American English words. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu), Brandon Rhodes (Linguist, Univ. of Chicago, Chicago, IL), Max Nelson, Kelly Berkson, and Kenneth de Jong (Linguist, Indiana Univ., Bloomington, IN)

A number of studies have shown that /r/ production in American English involves complex tongue shapes. Previous 3D imaging studies, however, have been limited to sustained (static) /r/ sounds produced in supine position. Since supine versus upright posture and static versus dynamic speech production influences the shape of the tongue, it is unclear to what degree previous findings of three-dimensional tongue shapes are generalizable to /r/ sounds produced dynamically and with upright posture. This study presents upright-posture 3D tongue shapes of dynamically produced /r/ sounds from words embedded in a carrier phrase. Twenty young adult native speakers of American English participated in the study (10 males and 10 females), and analyses are ongoing.

5aSCc19. Articulatory targets for ultrasound biofeedback determined by tracking regional tongue displacements. Sarah M. Hamilton, T. Douglas Mast, Michael Riley, and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3433 Clifton Ave., Cincinnati, OH 45220, hamilsm@mail.uc.edu)

Ultrasound biofeedback therapy (UBT) is a significant alternative to traditional therapy for speech disorders, but some users make little progress with the standard feedback display. For segments such as American English /r/, the required tongue movements are so complex that an ultrasound image of the tongue cannot adequately guide speakers to make the right movements. We propose to develop visual display regimes for ultrasound biofeedback that maximize speaker learning. As a preliminary step, we determine displacement ranges of tongue regions (blade, root, and dorsum) known to characterize accurate /r/ production, which will ultimately be used with robust image processing techniques to drive an improved UBT feedback display. Though blade, root, and dorsum movement are known to characterize /r/ production (Boyce *et al*., 2011; Espy-Wilson *et al*., 2000; Zhou *et al*., 2008), excursion of these tongue parts differs across individuals. Here, regional tongue displacements were tracked based on measured motion of local brightness maxima in low-pass-filtered midsagittal ultrasound images, normalized by reference anatomical landmark distances. Preliminary results show that the midsagittal dorsum, root, and blade show more extreme displacements relative to each other for the correct /r/ versus misarticulated /r/. This result is consistent with clinical observations of "humped" error /r/ tongue shapes.

5aSCc20. Models and methods for exploring anisotropy and inhomogeneity in vibrating vocal fold tissue. Ryan M. Oakey and Scott L. Thomson (Dept. of Mech. Eng., Brigham Young Univ. - Idaho, BYU-Idaho, Rexburg, ID 83460, rya07002@byui.edu)

Synthetic vocal fold models have long been used to study features of human voice production that cannot reasonably be studied *in vivo*. Numerous models have been developed to study different aspects of voice production. Some models have focused, for example, on mimicking geometry or tissue layering while others have been used to study the effect of vibration on live cells. Two aspects of vocal fold structure that have yet to be fully explored are the anisotropic and the inhomogeneous qualities of vocal fold tissue. These features significantly influence vocal fold flow-induced vibratory characteristics. The purpose of this research is to develop and explore models that incorporate anisotropy and inhomogeneity to study their effects on vocal fold vibration. In this presentation, methods and models for studying these features will be discussed; one example is the creation of a fiber matrix via rotary jet spinning that is embedded within silicone vocal fold models to study material anisotropy. Fabrication methodology and model response results will be presented, along with corresponding implications and possibilities for future use in the study of human voice production.

5aSCc21. Asymmetry in incomplete neutralization. Abby Kaplan (Linguist, Univ. of Utah, LNCO, Rm. 2300, 255 S Central Campus Dr., Salt Lake City, UT 84112, abby.kaplan@utah.edu)

Neutralizing patterns, such as final devoicing, are known to often be incomplete: "devoiced" final obstruents in languages such as Afrikaans may retain, e.g., longer preceding vowels than their voiceless counterparts. A common explanation for this phenomenon is that these partially devoiced obstruents are influenced by paradigmatically related obstruents that are fully voiced (e.g., Afrikaans *hoe[t]* ~ *hoe[d]e*, "hat(s)"). This proposal raises the question of whether such influence goes in the other direction too: is the [d] of *hoe[d]e* slightly devoiced, compared to a non-alternating [d] as in *roe[d]e* ("rod")? The experiment presented here tests this hypothesis in 28 Afrikaans nouns as produced by nine native speakers; vowel length, closure duration, release duration, and glottal pulses were measured as cues for voicing. The results provide no evidence for partial devoicing of voiced stops under the influence of devoiced counterparts elsewhere in the paradigm; that is, the [d] of *hoe[d]e* is no less voiced than the [d] of *roe[d]e*. I conclude that incomplete neutralization is an asymmetrical phenomenon: segments subject to neutralization, as in *hoe[t]*, may retain some contrasting cues; but these segments do not in turn encourage partial neutralization in morphologically related forms such as *hoe[d]e*.

5aSCc22. Efficiency of synthetic excitation obtained by interference of ultrasonic waveforms for reducing background noise for laryngeotomee. Romilla M. Bhat (Dept. of Electronics, Govt. Gandhi Memorial Sci. College, 9/C Om Nagar Udeywalla, Jammu, Jammu and Kashmir 180002, India, romillarosette@gmail.com) and Parveen K. Lehana (Dept. of Phys. and Electronics, Univ. of Jammu, Jammu, India)

Interference pattern of simulated ultrasonic waves using MATLAB(free version R2010a) using the laws of acoustics has been used to get excitation in the audio frequency range. Preliminary experiments have been done using different high frequencies and the recordings have been done in Goldwave 5.1 version and then subsequent analysis have been done in PRAAT to analyze the beat frequency obtained by interfering two high frequency (above 15 kHz) waves. The output thus obtained has been then subjected to Hilbert Transform for envelope detection using MATLAB(free version R2010a) to get the audio excitations. These audio excitations are further articulated by the glottal passage for speech synthesis for laryngeotomee or for producing alaryngeal speech. The results of simulation have been further utilised to fabricate a electronic circuit for generation of two continuous ultrasonic waves which are focussed to interfere inside the glottal tract, thereby producing a beat frequency in the audible range and reduce background noise. Index Terms: speech synthesis, ultrasonic waves, laryngeotomee Hilbert transform

5aSCc23. Testing a conceptual model of vocal tremor: Respiratory and laryngeal contributions to acoustic modulation. Jordon LeBaron and Julie Barkmeier-Kraemer (Univ. of Utah, 390 S. 1530 E., Rm. 1201 BEH SCI, Salt Lake City, UT 84112, jordon.lebaron@utah.edu)

Vocal tremor is a voice disorder characterized by rhythmic modulation of fundamental frequency (f_0) and sound pressure level (SPL) during sustained phonation. To date, contributions of oscillating speech structures to the acoustic modulations of vocal tremor are absent in the literature and are important for treatment purposes. The purpose of this study was to prospectively test the contributions of laryngeal and respiratory structure oscillations to acoustic modulation using singers to simulate vocal tremor. Laryngeal oscillation during production of natural vibrato was hypothesized to associate with f_0 modulation, whereas respiratory system oscillation during respiratory accented voicing was hypothesized to associate with SPL modulation. Ten female singers were recruited between 40 and 65 years of age without voice problems and meeting inclusion criteria for expertise necessary to produce natural vibrato and respiratory accented voicing during sustained phonation. Simultaneous endoscopic views of the larynx, respiratory kinematic, and acoustic signals were recorded during three trials of sustained /i/ during natural vibrato, or respiratory accented voicing. The rate and magnitude of kinematic movements of the chest wall and larynx are currently being compared to rate and magnitude of acoustic modulation. Preliminary qualitative evaluation of data supports hypothesized contributions of speech structures to acoustic modulation patterns.

5aSCc24. Normalizing nasality? Across-speaker variation in acoustical nasality measures. Will Styler (Dept. of Linguist, Univ. of Michigan, 611 Tappan St., 440 Lorch Hall, Ann Arbor, MI 48109, wstyler@umich.edu)

Although vowel nasality has traditionally been investigated using articulatory methods, acoustical methods have been increasingly used in the linguistic literature. However, while these acoustical measures have been shown to be correlated with nasality, their variability across-speakers is less-well investigated. To this end, we examined cross-speaker variation in coarticulatory vowel nasality in a large collection of multi-speaker English data, comparing measurements in CVC and NVN words at two points per vowel. Two known correlates were analyzed: A1-P0, where A1 is the amplitude of the harmonic under F1, and P0 is the amplitude of a low-frequency nasal peak (Chen 1997), and the bandwidth of F1 (Hawkins and Stevens 1985). Speakers varied both in terms of baseline measurements (e.g., the mean measurements in oral versus nasal vowels), and in the oral-nasal range (the measured difference between oral and nasal). This suggests that although analyzing centered within-speaker differences is unproblematic, the comparison of raw nasality measurements across speakers is unlikely to yield meaningful data. Moreover, this suggests that the perception of vowel nasality may require some process of speaker normalization, much like other aspects of vowel perception. Some suggestions for normalizing nasality measurements in research will be offered.

5aSCc25. A study on drinking judgment using phonation characteristics in speech signal. Won-Hee Lee and Myung-Jin Bae (SoongSil Univ., Hyungnam Eng. Bldg. 1212, Soongsil University 369 Sangdo-Ro, Dongjak-Gu, Seoul 156-743, South Korea, vbluelovev@ssu.ac.kr)

In this study, speech characteristics from before and after alcohol intoxication have been comparatively analyzed through speech analysis to obtain the degree of intoxication along with its parameters. The speech characteristics of an alcohol intoxicated person, such as pitch and formant and changing of sound levels, have been studied, but the speech characteristics before and after alcohol intoxication cannot be analyzed because of the sensitiveness of environmental changes. Therefore, we need to find more precise parameters, especially Phonation threshold pressure, the high levels of which result in dehydration of the laryngeal mucous membrane after alcohol intoxication. Speech after alcohol intoxication is with low lung pressure and decreasing lung capacity. Thus, we study the speech characteristics from before and after alcohol intoxication using speech-rate in speech signal and lung capacity.

FRIDAY MORNING, 27 MAY 2016 SALON G, 8:00 A.M. TO 11:45 A.M.

Session 5aSP

Signal Processing in Acoustics: General Topics in Signal Processing

Edmund Sullivan, Cochair
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Contributed Papers

8:00

5aSP1. Characterizing nonlinear systems with memory while combating and reducing the curse of dimensionality using new Volterra Expansion Technique. Albert H. Nuttall (Sonar and Sensors Dept., NUWCDIVNPT (retired), Old Lyme, CT), Derke R. Hughes, Richard A. Katz, and Robert M. Koch (NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@verizon.net)

A generalized model for characterizing nonlinear systems was originally proposed by Italian mathematician and physicist Vito Volterra (1860-1940). A further developed by American mathematician and MIT Professor Norbert Wiener (1894–1964) and published in 1958. After direct involvement with Norbert Wiener publication, Albert H. Nuttall has recently made new inroads along with his coauthors in applying the Wiener-Volterra model. A general description of a nonlinear system to the second order is termed the Nuttall-Wiener-Volterra model (NWV) after its co-founders. In this formulation, two measurement waveforms on the system are required in order to characterize a specified nonlinear system under consideration: an excitation input, x(t) (the transmitted signal) and a response output, z(t) (the received signal). Given these two measurement waveforms for a given system, a kernel response, h= [h0,h1,h2,h3] between the two measurement points, is computed via a least squares approach that optimizes modeled kernel values by performing a best fit between a measured response z(t) and a modeled response y(t). New procedures developed by A. Nuttall are invoked to significantly diminish the exponential growth of the computed number of kernel coefficients with respect to third order and higher orders to combat and reasonably reduce the curse of dimensionality.

8:15

5aSP2. Wigner cross-products encode relative phases. James B. Lee (None, 6016 S. E. Mitchell, Portland, OR 97206, cadwal@macforcego.com)

Two distinct kinds of elements comprise the Wigner time-frequency distribution. First are elements occurring at the actual times and frequencies present in the signal: call these “actual” elements. Second are elements occurring at the arithmetic mean of the actual elements: call these “ephemeral” elements. The actual elements essentially are a power spectrum resolved in time and frequency; always they are positive. The ephemeral elements, sometimes called “cross-products,” can be positive, negative, both, or zero, depending on the relative phases of the actual elements; they encode phase information necessary to complete time-frequency representation of all information in the original signal.

8:30

5aSP3. The blind source separation and classification of diver's noise based on adaptive beamforming and frequency selection. Juan Yang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, yang_juanacoustics@163.com), Karim G. Sabra (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Feng Xu, Xudong An, Hao Tang, Lei Jiang, and Peng Li (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

This paper introduces an algorithm to separate and classify the diver's noise in the presence of other noise interferers. The passive detection and classification for divers has been discussed before by using the breathing rate as features. But the performance of this approach significantly diminishes when other noise sources existing simultaneously, especially the other divers or louder shipping noises. In this paper, the wideband MVDR is used as an adaptive beamformer for horizontal linear array to better discriminate the azimuth of diver source of interest and separate the noises from different sources. The frequency content of the diver noise is also estimated adaptively according to azimuth distribution with no prior knowledge to enhance the separation ability from noises of similar azimuth and used to improve the correct rate of the classification. The performance of the proposed algorithm is discussed using both diver noise recorded in tank and at-sea shipping noise recorded in ocean for varying signal to noise ratio.

8:45

5aSP4. Impulse suppression algorithm development of a compatible program for cochlear implant users. Juliana Saba (Bioengineering, Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, jns109020@utdallas.edu), Jaewook Lee, Hussnain Ali (School of Behavioral and Brain Sci. (Speech & Hearing), Univ. of Texas at Dallas, Richardson, TX), Son Ta, Tuan Nguyen (Bioengineering, Univ. of Texas at Dallas, Richardson, TX), and John H. Hansen (School of Behavorial and Brain Sci. (Speech & Hearing), Univ. of Texas at Dallas, Richardson, TX)

Cochlear implant patients are more likely to abstain from impulsive-like environments due to limitations of current sound processing strategies. Conventional sound processing algorithms in conjunction with automatic gain controls tend to suppress impulsive sounds achieved unfortunately at the cost of reducing speech intelligibility. In this work, we propose a signal processing algorithm to provide protection to cochlear implant users in impulsive/noisy environments. A 22-band Advanced Combination Encoder (ACE) sound processing strategy was used as the base algorithm for speech enhancement. We propose an adaptable mathematical relationship to define impulsive like sound conditions and use this relationship to reduce the intensity of electrical pulses without altering the frequency ranges associated with speech to maintain speech quality/intelligibility. By varying the floor level of the environment, the program reduced/increased suppression intensity to achieve an effective configuration for practical listening environments. The algorithm was implemented in MATLAB and evaluated in benchtop (offline) testing with one CI user. Qualitative and quantitative analysis were performed using a paired preference test, a quality test, and an intelligibility test. Results showed increased quality of sound by +18%, and an average quality rating of processed speech 6/10 compared against 2.67/10 of speech in an unprocessed impulsive environment.

9:00

5aSP5. Bearing estimation using a single moving acoustic vector sensor. Edmund Sullivan (EJS_Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871, bewegungslos@fastmail.fm)

It is well known that a single moving pressure sensor can be used to estimate the bearing to a pure tone source if the source frequency is known a priori. It can also be shown, from the analysis of the relevant Grammian, that even if the source frequency and bearing are jointly estimated, say with a Kalman filter, it cannot be done, i.e., at least two sensors are necessary. However, if a PV sensor is used, additional measurements are provided, thereby allowing the estimation to be done. Although additional measurements are provided, there is still only a single physical sensor. Thus, even for small moving undersea vehicle, with its small amount of real estate, bearing measurements can be made. An example using simulated data is given.

9:15

5aSP6. Estimating blood pressures based on directly measured heart sounds. Lingguang Chen, Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, lingguang.chen@wayne.edu), Yong Xu (Dept. of Electric and Comput. Sci. Eng., Wayne State Univ., Detroit, MI), William D. Lyman, and Gaurav Kapur (Dept. of Pediatrics, Wayne State Univ., Detroit, MI)

The current standard technique for blood pressure determination is to use cuff/stethoscope, which is not suited for infants or children. Even for adults this approach yields 60% accuracy with respect to intra-arterial blood pressure measurements. Moreover, it does not allow for a continuous monitoring of blood pressures over the course of 24 h and even days. In this paper, a new methodology is developed that will enable one to estimate systolic and diastolic blood pressures based on the heart sounds measured directly from a human body. To this end, we must extract and separate the aortic, pulmonic, mitral, and tricuspid sounds that are involved in the first and second hear sounds, respectively, from the directly measured signals. Since many parameters such as the locations from which heart sounds are originated, and speed at which heart sounds travel inside a human body are unknown a priori, it is impossible to develop an analytic model to determine the blood pressure. Hence, an empirical model is developed to estimate blood pressures based on the data measured by two sensors. Preliminary clinical tests show that the methodology can separate heart sounds and estimated systolic and diastolic blood pressures correlate well with the benchmark data.

9:30

5aSP7. Marine mammal range and depth estimation using non-separated multipath propagation. Amelie Barazzutti (ASC/ENV, DGA Eng. and Integration, 60, bd du general Martial Valin, CS 21623, Paris Cedex 15 75509, France, amelie.barazzutti@gmail.com), Cédric GERVAISE (Chorus Chair, Fondation Grenoble-INP, Grenoble, France), Kerri D. SEGER, Aaron THODE (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA, U.S. Minor Outlying Islands), Jorge Urbán R., Pamela Martínez-Loustalot, M. Esther Jiménez-López, Diana López-Arzate (Universidad Autónoma de Baja California Sur, La Paz, Mexico), and Jérôme I. MARS (Gipsa-lab, Univ. Grenoble Alpes, Grenoble, France)

In passive acoustic monitoring, source localization using multipath propagation can be challenging whenever the source-receiver configuration leads to unresolved paths. Here, we propose a single-hydrophone method to estimate the range of a source based on two steps. First, we define a time deformation operator (e.g., Bonnel *et al.* 2014) to estimate the signal's impulse response (IR). Each group of paths, when warped at its time of arrival, is transformed into a tone that corresponds to a peak in the IR, which can be isolated in the warped time-frequency domain, permitting a significant improvement of the time resolution of the multipath, or groups of multipath. Next, relative arrival times are used to estimate the source range and depth. When only groups of paths are resolved, the identity of the paths constituting the group remains unknown, preventing the use of classical localization methods. To overcome this critical limitation, we apply an approach based on the characterization of groups of paths to estimate source location. We evaluate the method's performance on synthesized data, and examine its potential use on humpback whale low-frequency modulated calls using data collected off Cabo San Lucas in 2013.

9:45–10:00 Break

10:00

5aSP8. Joint beamforming and spectral enhancement for robust automatic speech recognition in reverberant environments. Fanuel Melak Asmare (Elec. Eng., Jacobs Univ. Bremen, Am Wall, 80, Bremen 28195, Germany, melakeegzi@gmail.com), Feifei Xiong (Fraunhofer IDMT, Oldenburg, Germany), Mathias Bode (Elec. Eng., Jacobs Univ. Bremen, Bremen, Germany), Bernard Mayer (Oldenburg Univ., Oldenburg, Germany), and Stefan Goetze (Fraunhofer IDMT, Oldenburg, Germany)

This work evaluates multi-microphone beamforming techniques and single-microphone spectral enhancement strategies to alleviate the reverberation effect for robust automatic speech recognition (ASR) systems in different reverberant environments characterized by different reverberation times T_{60}

and direct-to- reverberation ratios (DRRs). The systems under test consist of minimum variance distortionless response (MVDR) beamformers in combination with minimum mean square error (MMSE) estimators. For the later, reliable late reverberation spectral variance (LRSV) estimation employing a generalized model of the room impulse response (RIR) is crucial. Based on the generalized RIR model which separates the direct path from the remaining RIR, two different frequency resolutions in the short time Fourier transform (STFT) domain are evaluated, referred to as short- and long-term, to effectively estimate the direct signal. Regarding to the fusion between the MVDR beamformer and the MMSE estimator, the LRSV estimator can operate either on the multi-channel observed speech signals or on the single-channel beamformer output. By this, in this contribution, four different combination system architectures are evaluated and analyzed with a focus on optimal ASR performance w.r.t. word error rate (WER).

10:15

5aSP9. Single hydrophone, passive ranging of tonal sources in shallow water using the waveguide invariant. Andrew Young and Andrew Harms (Elec. and Comput. Eng., Duke Univ., 2127 Campus Dr., Durham, NC 27708, ayoung.ece@duke.edu)

A statistical method is presented for estimating both the value of the waveguide invariant, β, and the range to a tonal source in shallow water environments. This method requires minimal *a priori* environmental knowledge. A maximum-likelihood estimate of β is obtained from spectral analysis of a received signal from a tonal source over a known track. The range to a separate tonal source transiting the same shallow-water area is then estimated through a similar maximum-likelihood method, with the additional requirement of a range rate estimate. A recently published technique for estimating range rate through analysis of interference patterns of cross-correlated received time series is compared to a grid-search technique, in the context of evaluating performance of the range estimator. Both simulated and experimental results are presented. Simulated data were obtained using Kraken normal mode code. Experimental results used data from the Swellex ’96 (SW96) experiment. The negative impact of incorrect estimates of source range rate, varying bathymetry, and interfering sources are discussed.

10:30

5aSP10. A discourse on the effectiveness of digital filters at removing noise from audio. Nicole Kirch and Na Zhu (Eng. Technol., Austin Peay State Univ., 601 College St., Clarksville, TN 37044, nkirch@my.apsu.edu)

In real world applications, audio signals become degraded by irregular signals called noise. Cell phone, hearing aid, and audio recording industries rely on noise reduction in their signal processing to maintain clarity and intelligibility. This research project compares various digital filtering algorithms in their ability to remove noise without damaging the desired audio signal. The chosen filters operate either in the time or frequency domains: low pass, high pass, band pass, and windowing. Prerecorded and preprocessed samples of voice and music are saturated with computer generated noise or a physical noise source, and the original audio signals are then compared to the noisy and filtered signals. Effectiveness and unique characteristics of the filters are recorded and quantized. This data can be used in designing future acoustic devices.

10:45

5aSP11. Efficient modeling of room acoustic impulse response by segmentation into early and late components. Sahar Hashemgeloogerdi and Mark F. Bocko (Elec. and Comput. Eng., Univ. of Rochester, 405 Comput. Studies Bldg., Rochester, NY 14627, shashemg@UR.Rochester.edu)

A Room Impulse Response (RIR) has a complex time-frequency structure that can be divided into a number of discrete early reflections followed by a reverberant tail. Many acoustic signal enhancement applications, such as dereverberation and room equalization, require simple yet accurate models to represent an RIR. Parametric modeling of RIR typically approximate the overall response for a given position of a source and receiver inside a room, without considering the fundamental distinction between early reflection (modal region) and late reverberation (statistical region) of an RIR. In this paper, the RIR is estimated in two steps: first, the early reflection component is modeled utilizing fixed pole, parallel orthonormal basis functions to represent the modal behavior of the room. The late reverberation component of an RIR is more statistical in nature with exponentially decaying envelope, exhibiting different decay time constants for different frequencies. Wavelet decomposition using a multi-rate analysis filter bank is applied to model this part of the RIR. An iterative method is used to estimate the parameters of the model from a measured target RIR. The proposed hybrid method provides an accurate representation of an RIR while requiring a smaller number of parameters in comparison to existing methods.

11:00

5aSP12. Acoustic trajectory corrections to improve mapping of moving sources. Benoit Oudompheng, Lucille Lamotte (MicrodB, Ecully, France), and Barbara Nicolas (Lyon Univ., Créatis, CNRS, 7 Ave. Jean Capelle, Villeurbanne Cedex 69621, France, barbara.nicolas@creatis.insa-lyon.fr)

In the context of pass-by experiments, moving-source mapping is classically performed using the extension of conventional beamforming to moving sources: beamforming-MS. The main issue of moving-source mapping is the knowledge of the trajectory of the moving vehicle that contains the sources. This trajectory is mandatory to use the beamforming-MS. Indeed, trajectory errors induce localization artifacts in beamforming results, which can degrade beamforming-MS performance and bias physical interpretations of the localization results. This paper suggests an original method to acoustically correct trajectography mismatches, using a first beamforming-MS computation and spatial correlations, in order to improve moving-source mapping results. Experimental results of aerial pass-by experiments demonstrate the improvements to the mapping results using this trajectory correction.

11:15

5aSP13. Intensity filtering of acoustic sensor signals for multi-target tracking. Sihan Xiong (Penn State Univ., State College, PA), Thyagaraju Damarla (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD, thyagaraju.damarla.civ@mail.mil), and Asok Ray (Penn State Univ., State College, PA)

This paper addresses tracking of closely spaced moving vehicles by using passive acoustic signals. A multiple signal classification algorithm has been applied to estimate the bearing of targets from acoustic sensor signals. Field data from several sensor arrays have been used to reduce estimation variance in intensity filtering, which is performed on the bearing data with a sequential Monte Carlo method. Performance of the algorithm in terms of estimation of number of targets and their tracks will be presented using actual field data.

11:30

5aSP14. Non-inertial low frequency vector sensor with torsional suspension. Dimitri Donskoy (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, ddonskoy@stevens.edu)

Conventional low frequency (from 10 Hz) velocity vector sensors utilize inertial approach using neutrally buoyant body with inertial sensor (accelerometer or moving coil geophone) inside. In order to achieve high sensitivity at the low frequencies these sensors must have relatively high inertia (mass). High mass leads to increased size (such as soccer ball) of the sensor in order to satisfy the condition of neutral buoyancy. The developed alternative non-inertial sensor utilizes torsional frictionless and extremely compliant suspension enabling very light and small, yet sensitive with low resonance sensor.

Session 5aUW

Underwater Acoustics: Underwater Noise

Stan E. Dosso, Chair
School of Earth & Ocean Sci., Univ. of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada

Chair's Introduction—8:30

Contributed Papers

8:35

5aUW1. Using ambient noise correlations for relocating drifting sensor networks. Brendan Nichols, James Martin (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30309, bnichols8@gatech.edu), Chris Verlinden (Sci., U.S. Coast Guard Acad., New London, CT), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

A drifting sensor network, such as hydrophones mounted on autonomous underwater vehicles or drifting buoys, can be used as a random volumetric array for locating underwater acoustic sources. However, navigational uncertainties of mobile platforms underwater limit the positioning accuracy of their acoustic sensors. Precise positioning is required to perform coherent processing for sound source localization using this random volumetric array. It has been shown that ambient ocean noise correlations between fixed receivers can provide additional inter-element distance information to perform array element self-localization [Sabra *et al.*, IEEE J. Ocean Eng., **30** (2005)]. An extension of this approach for an array of drifting sensors will be proposed by optimizing the required averaging duration to account for sensor drift motion. Performance of this proposed approach will be demonstrated using at-sea data collected in the Long Island Sound.

8:50

5aUW2. Soundscape analysis in the Southern Ocean using elephant seals as acoustic glider of opportunity. Dorian Cazau, Julien Bonnel (Lab-STICC (UMR CNRS 6285), ENSTA Bretagne,Université Européenne de Bretagne, 2 rue François Verny, 29806 Brest Cedex 09, France, 2, rue François Verny, Brest 29200, France, julien.bonnel@ensta-bretagne.fr), Yves Le Bras, and Christophe Guinet (CEBC, UMR 7273 ULR-CNRS, Villiers en Bois, France)

The underwater ambient sound field contains quantifiable information about the physical and biological marine environment. Since 2011, we have been annually collecting underwater data over the migratory routes of bio-logged Southern Elephant Seal (SES). As done with classical underwater gliders, we extract from these data very high resolution (approximately 30 min/400 m) ocean ambient noise measurements. In this conference, we present an overall picture of the low-to-medium frequency (10–6000 Hz) ambient noise distribution and its variability in time and space at a regional scale within the Indian Ocean. We detail our methodology to extract robustly the measurements usually performed on ocean ambient noise, such as sound pressure level over different frequency bands and their statistical percentiles. Also, we present our first attempts of exploiting acoustic recordings from bio-logged SES to infer surface wind speed. Wind maps from the ASCAT satellite (IFREMER, France) were used to study correlation relations between surface wind speed and acoustic content (e.g., the ratio of sound pressure levels at 1 and 6 kHz). In complement, we test SVM and Neural Network methods to estimate the presence of different classes of winds (e.g., below and above 10 m/s) from underwater ocean noise.

9:05

5aUW3. Identifying frequency and temporal characteristics of ambient noise using correlation matrices. Stephen Nichols and David L. Bradley (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu)

Long-term ambient noise recordings contain an immense amount of information which quickly becomes cumbersome to analyze fully and efficiently. Correlation matrices provide a promising approach to identifying and isolating the active source mechanisms in ambient noise datasets. By comparing the noise levels in different frequency bands, frequency regions which tend to change together are identified, pointing toward either one common source or multiple related sources for the identified frequency bands. An attempt at simplifying the frequency and temporal information provided by series of correlation matrices is presented, utilizing data recorded by the Comprehensive Nuclear-Test Ban Treaty Organization's hydroacoustic monitoring system.

9:20

5aUW4. Time-domain cross-correlation using a normal mode formulation in shallow water. Haiqiang Niu and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., San Diego, CA 92093-0238, hniu@ucsd.edu)

Cross-correlation of ambient noise fields can yield temporal peaks, from which the travel time of acoustic paths between the sensors can be estimated. A good approximation of the time arrival structure of the actual Green's function is obtained if the noise field is isotropic. However, this condition is rarely satisfied in the ocean environment. For instance, ship noise is a common discrete source. Typically, the existence of discrete sources can affect the ambient noise cross-correlation. This paper presents a time-domain normal mode formulation of cross-correlation function for a single point source in a shallow water environment. The relation between cross-correlation function and time-domain Green's function is investigated for the point source by using time-domain normal mode formulations. The cross-correlation of the same modes yields two-point Green's function provided that the point source is in the end-fire direction, while the cross-correlation of the modes with different indexes represents a range-dependent component.

9:35

5aUW5. Passive characterization of underwater sound channel using ray-based blind deconvolution algorithm. Sung-Hoon Byun (KRISO, 32 1312 Beon-gil, Yuseong-daero, Yuseong-gu, Daejeon 34103, South Korea, byunsh@kriso.re.kr) and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Ray-based blind deconvolution (RBD) algorithm is a blind channel estimation method, which can estimate both the channel impulse response and the original signal broadcast from an unknown source. It is based on high-

frequency ray assumption and requires only the elementary information of array geometry and sound speed at the array position. Its capability of estimating the channel in a passive manner can be used to infer the sound channel characteristics from a source of opportunity such as passing surface ship whose positioning information is available from geo-referencing systems like the Automatic Identification System. Recently, we applied the RBD algorithm to the shipping noise data recorded on vertical line arrays at multiple locations in shallow water and confirmed its applicability for channel estimation and performing environmental inversion. This presentation shows the channel impulse responses estimated from a variety of source-receiver geometry and discusses the characteristics of sound propagation as well as the environmental property such as bottom loss which can be observed using the RBD algorithm.

9:50–10:05 Break

10:05

5aUW6. Detecting ice noise in Arctic ambient noise recorded on a drifting vertical line array. Emma Reeves, Peter Gerstoft, Peter Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu)

From April to September 2013, a 600-m Distributed Vertical Line Array (DVLA) with 22 hydrophones drifted from the North Pole toward the Fram Strait. The DVLA recorded low-frequency ambient noise (1953.125 Hz sampling) for 108 minutes six days per week. Spectral analysis of the ambient noise data indicates the presence of acoustic transients, which include airgun and drilling sources as well as ice shear-induced broadband events and FM tones. Similar ice shear signatures in the Beaufort Sea were highly correlated with environmental variables leading to rifting [Kinda *et al.*, J. Acoust. Soc. Am. **138**, 2034–2045 (2015)], demonstrating the potential of acoustics to monitor rapid sea ice processes and pack ice breakup. Motivated by this study, a wavelet-based audio fingerprinting method is applied for identifying ice noise in the drifting DVLA data. The algorithm blindly groups spectral signatures, allowing the categorization of previously unrecognized signatures. The size of each category at points along the drift path is presented as a local, seasonally dependent estimate of the presence of ice noise.

10:20

5aUW7. Very high frequency underwater noise from breaking waves. Grant B. Deane and M. D. Stokes (Marine Physical Lab., Scripps Inst. of Oceanogr., Code 0206, UCSD, La Jolla, CA 92093-0238, gdeane@ucsd.edu)

In the absence of anthropogenic and biological sources and under wind-driven seas, underwater sound generated by breaking waves is the dominant source of high frequency ambient noise in the ocean. At frequencies above O (1000 Hz), this noise is generated by pulses of sound radiated by air bubbles entrained in actively breaking whitecaps. A history of measurements extending back more than 50 years has shown that the sound radiated by breaking waves is well-correlated with wind speed and extends in frequency to at least 20 kHz. Here, we present measurements of high frequency noise radiated by breaking laboratory waves. Focused wave packets in the Glass Wave Channel at the Hydraulics Laboratory at SIO generated breaking waves ranging in type from spilling to plunging, which were representative of small to medium scale waves on seas driven by 8–12 m/s winds. Measurements made with high frequency hydrophones (ITC 1089D) show that the noise spectrum from these waves extends beyond 400 kHz, challenging the assumption that noise above 100 kHz is dominated by thermal effects. Moreover, the slope of the noise spectrum is roughly −17 dB/decade, consistent with oceanic noise spectra at lower frequencies.

10:35

5aUW8. Sound production in two species of eelgrass. Jeff Rice (Univ. of Washington, 18529 29th Ave. NE, Lake Forest Park, WA 98155, jrice1000@mac.com)

There have been numerous anecdotal reports that a species of the aquatic plant known as eelgrass (*Zostera marina*) makes an audible bubbling sound that can be heard on warm, sunny days in estuarine waters at low tide. This champagne-like fizzing sound results from the expiration of oxygen by the plant during photosynthesis and can be heard as airborne sound by listening closely to the surface of the water. The author recorded this sound in July of 2014 in the shallow mudflats of Puget Sound in Washington. Recordings were made of both airborne and underwater sounds. On the same day in the same general area, the author also recorded similar bubbling from a different, non-native species of eelgrass (*Zostera japonica*). The sounds of the two species were compared, revealing clear and distinct sound signatures for each. These recordings suggest that sound production in the two species of eelgrass is strongly present and may provide opportunities for further research.

10:50

5aUW9. An adaptive signal tools design inspired by marine mammals calls. Siyuan Cang and Xueli Sheng (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong Str., Nangang Dist., Harbin 150001, China, cangsiyuan@hrbeu.edu.cn)

As the main wisdom animal in the ocean, the marine mammals like dolphins and whales has got excellent capabilities for complex ocean environments through millions of years of evolution. They physically look simple, but are able to do lots of fantastic jobs, such as echolocation, preying, and even communication among populations of marine mammals. This paper proposes a method of designing an active sonar waveform inspired by the marine mammals calls. Two math models of marine mammal sound signal are adopted, one AR model, the other piecewise linear chirp model. The time-frequency structures, ambiguity function and Q function will be analyzed. As the simulation results shown, it has a good range and velocity resolution under environment with reverberation. In the end, a concept named Marine Mammal Signal Tools (MMST) is presented. The signal generating from MMST can be adaptive to varied ocean environments. In the sense, it can be extensively used and plays more significant roles in the modern sonar.

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Information for contributors to the Journal of the Acoustical Society of America (JASA)

Editorial Staff[a)]

Journal of the Acoustical Society of America, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300

The procedures for submitting manuscripts to the *Journal of the Acoustical Society of America* are described. The text manuscript, the individual figures, a reviewer PDF, and a cover letter are each uploaded as separate files to the *Journal's* Manuscript Submission and Peer Review System. The required format for the text manuscript is intended so that it will be easily interpreted and copy-edited during the production editing process. Various detailed policies and rules that will produce the desired format are described, and a general guide to the preferred style for the writing of papers for the *Journal* is given. Criteria used by the editors in deciding whether or not a given paper should be published are summarized.

TABLE OF CONTENTS

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I. INTRODUCTION

The present document is intended to serve jointly as (i) a set of directions that authors should follow when submitting articles to the *Journal of the Acoustical Society of America* and as (ii) a style manual that describes those stylistic features that are desired for the submitted manuscript. Authors may refer to recent issues of the *Journal* for examples of how specific style issues are handled.

II. ONLINE HANDLING OF MANUSCRIPTS

All new manuscripts intended for possible publication in the *Journal of the Acoustical Society of America* should be submitted online. The steps involved in the processing of manuscripts that lead from the initial submission through the peer review process to the transmittal of an accepted manuscript to the production editing office are handled by a computerized system referred to here as the Editorial Manager (EM) system. The Acoustical Society of America contracts with Aries Systems, Inc. for the use of this system. There is one implementation that is used for most of the material that is submitted to the *Journal of the Acoustical Society of America* (JASA) and a separate implementation for the special section *JASA Express Letters (JASA-EL)* of the *Journal.*

A. Registration

Everyone involved in the handling of manuscripts in the *Journal's* editorial process must first register with the *Journal's* implementation of the EM system, and the undertaking of separate actions, such as the submission of a manuscript, requires that one first log-in to the system at www.editorialmanager.com/JASA or www.editorialmanager.com/JASA-EL.

If you have never logged into the system, you will need to get a user name and password. Many ASA members are already in the database, so if you are a member, you in principle may already have a user name and password.

Once you have your "user name" and "password" you go to the log-in page, and give this information when you log-in.

B. Overview of the editorial process

(1) An author denoted as the corresponding author submits a manuscript for publication in the *Journal.*
(2) A quick quality control check is done on the manuscript. If there are too many (n>15) errors in the submitted manuscript, it will be returned to the corresponding author to fix them and resubmit.
(3) One of the *Journal's* Associate Editors is recruited to handle the peer-review process for the manuscript.
(4) The Associate Editor recruits reviewers for the manuscript via the online system.
(5) The reviewers critique the manuscript, and submit their comments online via the EM system.
(6) The Associate Editor makes a decision regarding the manuscript, and then composes online an appropriate decision letter, which may include segments of the reviews, and which may include attachments.
(7) The *Journal's* staff transmits a letter composed by the Associate Editor to the corresponding author. This letter describes the decision and further actions that can be taken.

If revisions to the manuscript are invited, the author may resubmit a revised manuscript, and the process cycle is repeated. To submit a revision authors should use the link provided in the decision message.

C. Preparation for online submission

Before one begins the process of submitting a manuscript online, one should first read the document *Ethical Principles of the Acoustical Society of America for Research Involving Human and Non-Human Animals in Research and Publishing and Presentations* which is reached from the site <http://scitation.aip.org/content/asa/journal/jasa/info/authors>. During the submission, you will be asked if your research conformed to the stated ethical principles and if your submission of the manuscript is in accord with the ethical principles that the Acoustical Society has set for its journals. If you cannot confirm that your manuscript and the research reported are in accord with these principles, then you should not submit your manuscript.

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    (b) E-mail address

(c) Work telephone number
(d) Postal address (required for corresponding author, otherwise optional)

(2) Title and running title of the paper. The running title is used as part of the table of contents on the journal cover. (The title is preferably limited to 17 words and the running title is limited to six words and up to 50 characters and spaces; neither may include non-obvious acronyms or any words explicitly touting novelty.)
(3) Abstract of the paper. (This must be in the form of a single paragraph and is limited to 200 words for regular articles and to 100 words for Letters to the Editor. (Authors would ordinarily do an electronic pasting from a text file of their manuscript.)
(4) Four keywords that characterize the subject matter of the paper.
(5) A short prioritized list of Associate Editors suggested for the handling of the manuscript. (EM currently limits to one, but that will be expanded.)
(6) Contact information (name, e-mail address, and institution) of suggested reviewers (if any), and/or names of reviewers to exclude.
(7) Cover letter file. This should supply additional information that should be brought to the attention of the editor(s) and/or reviewer(s).
(8) Properly prepared manuscript/article file in LaTeX or Word format. (The requirements for a properly prepared manuscript are given further below.) It must be a single stand-alone file. If the author wishes to submit a LaTeX file, the references should be included in the file, not in a separate BibTeX file. Authors should take care to ensure that the submitted manuscript/article file is of reasonable length.
(9) Properly prepared figure files in TIFF, PS, JPEG, or EPS (see also Section V. H); one file for each cited figure number. The uploading of figures in PDF format is not allowed. (The captions should be omitted, and these should appear as a list in the manuscript itself.) The figures should not have the figure numbers included on the figures in the files. Authors may upload figures in a zip file. For figures without subparts (as well as for figures having subparts built in to a single file), the uploaded file should be named "Figure1.nnn" where "nnn" is the type of graphic file (.jpg, .eps, etc.). For compound figures uploaded as separate files, the individual files should be named Figure1a.nnn, Figure1b.nnn, etc., where "nnn" is the correct filetype/file extension. In order to maintain online color as a free service to authors, the *Journal* cannot accept multiple versions of the same file. Authors may not submit two versions of the same illustration (e.g., one for color and one for black & white). When preparing illustrations that will appear in color in the online *Journal* and in black & white in the printed *Journal,* authors must ensure that: (i) colors chosen will reproduce well when printed in black & white and (ii) descriptions of figures in text and captions will be sufficiently clear for both print and online versions. For example, captions should contain the statement "(Color online)." If one desires color in both versions, these considerations are irrelevant, although the authors must guarantee that mandatory additional publication charges will be paid.
(10) Supplemental files (if any) that might help the reviewers in making their reviews. If the reading of the paper requires prior reading of another paper that has been accepted for publication, but has not yet appeared in print, then a PDF file for that manuscript should be included as a supplementary file. Also, if the work draws heavily on previously published material which, while available to the general public, would be time-consuming or possibly expensive for the reviewers to obtain, then PDF files of such relevant material should be included.
(11) Archival supplemental materials to be published with the manuscript in AIP Publishing's Supplemental Materials electronic depository.

In regard to the decision as to what formats one should use for the manuscript and the figures, a principal consideration may be that the likelihood of the published manuscript being more nearly to one's satisfaction is considerably increased if AIP Publishing, during the production process, can make full or partial use of the files you submit. There are conversion programs, for example, that will convert LaTeX and MS Word files to the typesetting system that AIP Publishing uses. If your manuscript is not in either of these formats, then it will be completely retyped. If the figures are submitted in EPS, PS, JPEG, or TIFF format, then they will probably be used directly, at least in part. The uploading of figures in PDF format is not allowed.

D. Steps in online submission

After logging in, one is brought to the EM author main page, and can select the option of submitting a new manuscript. The resulting process leads the corresponding author through a sequence of screens.

The submit screen will display a series of fairly self-explanatory tabs. Clicking on these tabs displays the tasks that must be completed for each step in the submission.

After submission, all of the individual files, text and tables, plus figures, that make up the full paper will be merged into a single PDF file. One reason for having such a file is that it will generally require less computer memory space. Another is that files in this format are easily read with any computer system. However, the originally submitted set of files, given the acceptance for publication, will be what is submitted to the Production Editing office for final processing.

E. Quality check by editorial office

Upon receiving system notification of a submission, staff members in the Editorial Office check that the overall submission is complete and that the files are properly prepared and suitable for making them available to the Associate Editors and the reviewers. If all is in order, the Manuscript Coordinator initiates the process, using the keywords and suggested Associate Editor list supplied by the author, to recruit an Associate Editor who is willing to handle the manuscript. At this time the author also receives a "confirmation of receipt" e-mail message. If the staff members deem that there are numerous or serious submission defects that should be addressed, then the author receives a "quality check" e-mail message. If there are only a small number of defects, the e-mail message may give an explicit description of what is needed. In some cases, when the defects are very numerous, and it is apparent that the author(s) are not aware that the *Journal* has a set of format requirements, the e-mail message may simply ask the authors to read the instructions (i.e., the present document) and to make a reasonable attempt to follow them.

III. PUBLICATION CHARGES

A. Mandatory charges

Papers of longer length or with color figures desired for the print version of the *Journal* will not be published unless it is first agreed that certain charges will be paid. If it is evident that there is a strong chance that a paper's published length will exceed 12 pages, the paper will not be processed unless the authors guarantee that the charges will be paid. If the paper's published length exceeds 12 pages or more, there is a mandatory charge of $80 per page for the entire article. (The mandatory charge for a 13 page article, for example, would be $1040, although there would be no mandatory charge if the length were 12 pages.)

Manuscripts should not exceed 10,500 words [approximately twelve (12) printed journal pages]. Abstract, title, author list, references, and acknowledgments are all excluded from the 10,500-word limit. Figures, tables, and equations, however, are included and must be accounted for by calculating a word count equivalent to the space they occupy. Circumvention of the length limitation is contrary to the purpose of this journal.

Please use these guidelines for estimating length.

TeX users

Authors are advised to use the article class of TeX. If the version of the manuscript obtained using the "reprint" option fits on twelve (12) pages with a font size of 12 points, the length should be acceptable.

Word users

Highlight the manuscript text, excluding abstract, author list, acknowledgments and references, and note the word count at the bottom of the screen. Add to that the word-count-equivalents for figures, tables, and equations as follows:

Figures: An average single-column figure will displace 220 words. For a more accurate estimation, use the following: 150/aspect ratio + 20 words for single-column figures and 300/0.5 x aspect ratio + 40 words for double-column figures. Aspect ratio = width/height.

Tables: 6.5 words per line, plus 13 words for single-column tables. 13 words per line, plus 26 words for double-column tables.

Equations: 16 words per row for single-column equations. 32 words per row for double-column equations.

If the total number of words (text + figures + tables + equations) is 10,500 or less, the length is acceptable.

Color figures can be included in the online version of the *Journal* with no extra charge, providing that these appear suitably as black and white figures in the print version.

The charges for inclusion of color figures in the print version of the *Journal* are $325 per figure file.

If an author's institution or research sponsor is unwilling to pay such charges, the author should make sure that all of the figures in the paper are suitable for black and white printing, and that the estimated length is manifestly such that it will not lead to a printed paper that exceeds 12 pages.

JASA now offers a "gold" open access option, the price of which is $2200 USD. If an open access paper runs beyond 12 pages, overpage fees are only due on the pages beyond 12.

B. Optional charges

To encourage a large circulation of the *Journal* and to allow the inclusion of a large number of selected research articles within its volumes, the *Journal* seeks partial subsidization from the authors and their institutions. Ordinarily, it is the institutions and/or the sponsors of the research that undertake the subsidization. Individual authors must ask their institutions or whatever agencies sponsor their research to pay a page charge of $80 per printed page to help defray the publication costs of the *Journal.* (This is roughly 1/3 of the actual cost per page for the publication of the *Journal.*) The institutions and the sponsoring agencies have the option of declining, although a large fraction of those asked do pay them. The review and selection of manuscripts for publication proceeds without any consideration on the part of the Associate Editors as to whether such page charges will be honored. The publication decision results after consideration of the factors associated with peer review; the acceptance of the page charges is irrelevant.

C. Waived charges

To encourage submission of review papers, tutorial papers, forum papers, and special external (to our specialties) papers, all of which are invited, we now waive the publication fee for these article types. However, a fee for (optional) color in the print version will be requested for such articles.

D. Payment of publication charges—RightsLink

When your page proofs are ready for your review, you will receive an e-mail from AIP Publishing Production Services. It will include a link to an online RightsLink site where you can pay your voluntary or mandatory page charges, color figure charges, or to order reprints of your article. If you are unable to remit payment online, you will find instructions for requesting a printed invoice so that you may pay by check or wire transfer.

IV. FORMAT REQUIREMENTS FOR MANUSCRIPTS

A. Overview

For a manuscript to pass the initial quality control, it is essential that it adhere to a general set of formatting requirements. Such vary from journal to journal, so one should not assume that a manuscript appropriate for another journal's requirements would be satisfactory for the *Journal* *of the Acoustical Society of America.* The reasons for the *Journal's* requirements are partly to ensure a uniform style for publications in the *Journal* and partly to ensure that the copy-editing process will be maximally effective in producing a quality publication. While some submitted papers will need very few or no corrections, there is a sufficiently large number of accepted papers of high technical merit that need such editing to make it desirable that all submissions are in a format that amply allows for this.

The following is a list of some of the more important requirements. (More detailed requirements are given in the sections that follow.)

(1) The manuscript must be paginated, starting with the first page.
(2) The entire manuscript must be doubled-spaced. This includes the author addresses, the abstract, the references, and the list of figure captions. It should contain no highlighting.
(3) The title and author list is on the first page. The abstract is ordinarily on a separate page (the second page) unless there is sufficient room on the title page for it, within the constrains of ample margins, 12 pt type, double-spacing, and ample white space. The introduction begins on a separate page following the page that contains the abstract.
(4) The title must be in lower case, with the only capitalized words being the first word and proper nouns.
(5) No acronyms should be in the title or the running title unless they are so common that they can be found in standard dictionaries.
(6) No unsupported claims for novelty or significance should appear in the title or abstract, such as the use of the words *new, original, novel, important,* and *significant.*
(7) The abstract should be one paragraph and should be limited to 200 words (100 words for Letters to the Editor).
(8) Major section headings should be numbered by capital roman numerals, starting with the Introduction. Text of such headings should be in capital letters.
(9) Reference citations should include the full article titles and page ranges of all cited papers.
(10) There should be no personal pronouns in the abstract.
(11) No more than one-half of the references should be to the authors themselves.
(12) The total number of figures should not ordinarily be more than 20 (see Section V. H).

B. PDF for reviewers

A PDF file with line numbers and inline figures and captions needs to be provided by the author(s) for each manuscript, for the ease of the reviewers.

C. Keyboarding instructions

Each submitted paper, even though submitted online, should correspond to a hard copy manuscript. The electronic version has to be prepared so that whatever is printed-out will correspond to the following specifications:

(1) The print-out must be single sided.
(2) The print-out must be configured for standard US letter paper (8.5″ by 11″).
(3) The text on any given page should be confined to an area not to exceed 6.5″ by 9″. (One inch equals 2.54 cm.) All of the margins when printed on standard US letter paper should be at least 1″.
(4) The type font must be 12 pt, and the line spacing must correspond to double spacing (approximately 1/3″ or 0.762 cm per line of print). The fonts used for the text must be of a commonly used easily readable variety such as Times, Helvetica, New York, Courier, Palatino, and Computer Modern.
(5) The authors are requested to use computers with adequate word-processing software in preparing their manuscripts. Ideally, the software must be sufficiently complete that all special symbols used in the manuscript are printed. (The list of symbols available to AIP Publishing for the publication of manuscripts includes virtually all symbols that one can find in modern scientific literature. Authors should refrain from inventing their own symbols.) It is preferred that vectors be designated by boldface symbols within a published paper rather than by arrows over the symbols.
(6) Manuscript pages must be numbered consecutively, with the title page being page 1.

D. Order of pages

The manuscript pages must appear in the following order:

(1) Title page. (This includes the title, the list of authors, their affiliations, an abbreviated title for use as a running title in the published version, and any appropriate footlines to title or authors.)
(2) Abstract page, which may possibly be merged with the title page if there is sufficient room. Please note that the *Journal* requires the abstract to be typed double spaced, just as for all of the remainder of the manuscript.
(3) Text of the article. This must start on a new page.
(4) Acknowledgments.
(5) Appendixes (if any).
(6) Textual footnotes. (Allowed only if the paper cites references by author name and year of publication.)
(7) References. (If the paper cites references by labeling them with numbers according to the order in which they appear, this section will also include textual footnotes.)
(8) Tables, each on a separate page and each with a caption that is placed above the table.
(9) Collected figure captions.

Figures should not be included in the "Article" file.

E. Title page of manuscript

The title page should include on separate lines, with appropriate intervening spacing: The article title, the name(s) of author(s), one complete affiliation for each author, and the date on which the manuscript is uploaded to the JASA manuscript submission system.

With a distinctive space intervening, the authors must give, on a separate line, a suggested running title of six words or less that contains a maximum of 50 characters. The running title appears on the front cover of the *Journal* as part of an abbreviated table of contents, and it is important that it give a nontrivial indication of the article's content, although some vagueness is to be expected.

Titles should briefly convey the general subject matter of the paper and should not serve as abstracts. The upper limit is set at 17 words. They must be written using only words and terminology that can be found in standard unabridged US English dictionaries or in standard scientific/technical dictionaries, and they must contain no acronyms other than those that can be found in such dictionaries. Ideally, titles should be such that they contain appropriate keywords. This will enable a reader doing a computer-assisted search to determine whether the paper has any relevance to a given research topic. Begin the first word of the title with a capital letter; thereafter capitalize only proper nouns. The *Journal* discourages the use of subjective words such as "original," "new," "novel," "important," and "significant" in the title. In general, words whose sole purpose is to tout the importance of a work are regarded as unnecessary; words that clarify the nature of the accomplishment are preferred.

In the list of authors, to simplify later indexing, adopt one form of each name to use on the title pages of all submissions to the *Journal.* It is preferred that the first name be spelled out, especially if the last name is a commonly encountered last name. If an author normally uses the middle name instead of the first name, then an appropriate construction would be one such as J. John Doe. Names must be written with last name (family name) given last. Omit titles such as Professor, Doctor, Colonel, Ph.D., and so on.

Each author may include only one current affiliation in the manuscript. Put the author's name above the institutional affiliation. When there is more than one author with the same institutional affiliation, put all such names above the stating of that affiliation. (See recent issues of the *Journal* for examples.)

In the stating of affiliations, give sufficient (but as briefly as possible) information so that the corresponding author may be contacted by mail and/or e-mail by interested readers. Do not give websites, telephone numbers, or FAX numbers. Names of states and countries should be written out in full. If a post office box should be indicated, append this to the zip code (as in 02537-0339). Use no abbreviations other than DC (for District of Columbia). If the address is in the United States, include "USA."

The preferred order of listing of authors is in accord with the extent of their contributions to the research and to the actual preparation of the manuscript. (Thus, the last listed author is presumed to be the person who has done the least.)

The stated affiliation of any given author should be that of the institution that employed the author at the time the work was done. In the event an author was employed simultaneously by several institutions, the stated affiliation should be that through which the financial support for the research was channeled. If the current (at the time of publication) affiliation is different, then that should be stated in a footline. If an author is deceased then that should be stated in a footline. (Footlines are discussed further below.)

There is no upper limit to the number of authors of any given paper. If the number becomes so large that the appearance of the paper when in print could look excessively awkward, the authors will be given the option of not explicitly printing the author affiliations in the heading of the paper. Instead, these can be handled by use of footlines as described below. The *Journal* does not want organizations or institutions to be listed as authors. If there are a very large number of authors, those who made lesser contributions can be designated by a group name, such a name ending with the word "group." A listing of the members of the group possibly including their addresses should be given in a footline.

Footlines to the title and to the authors' names are consecutively ordered and flagged by lowercase alphabetical letters, as in Fletcher[a)], Hunt[b)], and Lindsay[c)]. If there is any history of the work's being presented or published in part earlier, then a footline flag should appear at the end of the title, and the first footline should be of the form exemplified below:[1]

[a)]Portions of this work were presented in "A modal distribution study of violin vibrato," Proceedings of International Computer Music Conference, Thessaloniki, Greece, September 1997, and "Modal distribution analysis of vibrato in musical signals," Proceedings of SPIE International Symposium on Optical Science and Technology, San Diego, CA, July 1998.

Authors have the option of giving a footline stating the e-mail address of one author only (usually the corresponding author), with an appropriate footline flag after that name and with each footline having the form:

[b)]Electronic mail: name@servername.com

F. Abstract page

Abstracts are often published separately from actual articles, and thus are more accessible than the articles themselves to many readers. Authors consequently must write abstracts so that readers without immediate access to the entire article can decide whether the article is worth obtaining. The abstract is customarily written last; the choice of what should be said depends critically on what is said in the body of the paper itself.

The abstract should not be a summary of the paper. Instead, it should give an accurate statement of the subject of the paper, and it should be written so that it is intelligible to a broad category of readers. Explicit results need not be stated, but the nature of the results obtained should be stated. Bear in mind that the abstract of a journal article, unlike the abstract of a talk for a meeting, is backed-up by a written article that is readily (if not immediately) accessible to the reader.

Limit abstracts to 200 words (100 words for Letters to the Editor). Do not use footnotes. If the authors decide that it is imperative to cite a prior publication in the abstract, then the reference should be embedded within the text and enclosed within square brackets. These should be in one of the two standard JASA formats discussed further below, but titles of articles need not be given. The abstract should contain no acknowledgments. In some circumstances, abstracts of longer than 200 words will be allowed. If an author believes that a longer abstract is essential for the paper, they should send an e-mail message to jasa@aip.org with the subject line "Longer abstract requested." The text of the desired abstract should be included in the memo, along with a statement of why the author believes the longer abstract is essential. The abstract will be reviewed by the editors, and possibly a revised wording may be suggested.

Personal pronouns and explicit claims as to novelty should be assiduously avoided. Do not repeat the title in the abstract, and write the abstract with the recognition that the reader has already read the title. Avoid use of acronyms and unfamiliar abbreviations. If the initial writing leads to the multiple use of a single lengthy phrase, avoid using an author-created acronym to achieve a reduction in length of the abstract. Instead, use impersonal pronouns such as *it* and *these* and shorter terms to allude to that phrase. The shortness of the abstract reduces the possibility that the reader will misinterpret the allusion.

G. Section headings

The text of a manuscript, except for very short Letters to the Editor, is customarily broken up into sections. Four types of section headings are available: principal heading, first subheading, second subheading, and third subheading. The principal headings are typed boldface in all capital letters and appear on separate lines from the text. These are labeled by uppercase roman numerals (I, II, III, IV, etc.), with the introductory section being principal section I. First subheadings are also typed on separate lines; these are labeled by capital letters: A, B, C, etc. The typing of first subheadings is boldface, with only the first word and proper nouns being capitalized. Second subheadings are ordered by numbers (1, 2, 3, etc.) and are also typed on separate lines. The typing of second subheadings is italic boldface, also with only the first word and proper nouns capitalized. Third subheadings appear in the text at the beginning of paragraphs. These are labeled by lowercase letters (a, b, c, etc.) and these are typed in italics (not boldfaced). Examples of these types of headings can be found in recent issues of the *Journal.* (In earlier issues, the introduction section was not numbered; it is now required to be numbered as the first principal section.)

Headings to appendixes have the same form as principal headings, but are labeled by uppercase letters, with an optional brief title following the identification of the section as an appendix, as exemplified below:

APPENDIX C: CALCULATION OF IMPEDANCES

If there is only one appendix, the letter designation should be omitted.

V. STYLE REQUIREMENTS

A. Citations and footnotes

Regarding the format of citations made within the text, authors have two options: (1) textual footnote style and (2) alphabetical bibliographic list style.

In the *textual footnote style,* references and footnotes are cited in the text by superscripted numerals, as in "the basic equation was first derived by Rayleigh[44] and was subsequently modified by Plesset.[45]" References and footnotes to text material are intercalated and numbered consecutively in order of first appearance. If a given reference must be cited at different places in the text, and the citation is identical in all details, then one must use the original number in the second citation.

In the *alphabetical bibliographic list style,* footnotes as such are handled as described above and are intended only to explain or amplify remarks made in the text. Citations to specific papers are flagged by parentheses that enclose either the year of publication or the author's name followed by the year of publication, as in the phrases "some good theories exist (Rayleigh, 1904)" and "a theory was advanced by Rayleigh (1904)." In most of the papers where this style is elected there are no footnotes, and only a bibliographic list ordered alphabetically by the last name of the first author appears at the end of the paper. In a few cases,[2] there is a list of footnotes followed by an alphabetized reference list. Within a footnote, one has the option of referring to any given reference in the same manner as is done in the text proper.

Both styles are in common use in other journals, although the *Journal of the Acoustical Society of America* is one of the few that allows authors a choice. Typically, the textual footnote style is preferred for articles with a smaller number of references, while the alphabetical bibliographic list style is preferred for articles with a large number of references. The diversity of the articles published in the *Journal* makes it infeasible to require just one style unilaterally.

B. General requirements for references

Regardless of what reference style the manuscript uses, the format of the references must include the titles of articles. For articles written in a language other than English, and for which the Latin alphabet is used, give the actual title first in the form in which it appeared in the original reference, followed by the English translation enclosed within parentheses. For titles in other languages, give only the English translation, followed by a statement enclosed in parentheses identifying the language of publication. Do not give Latin-alphabet transliterations of the original title. For titles in English and for English translations of titles, use the same format as specified above for the typing of the title on the title page. Begin the first word of the title with a capital letter; thereafter capitalize only those words that are specified by standard dictionaries to be capitalized in ordinary prose.

One must include only references that can be obtained by the reader. One should also not cite any paper that has only been submitted to a journal; if it has been accepted, then the citation should include an estimated publication date. If one cites a reference, then the listing must contain enough information that the reader can obtain the paper. If theses, reports, or proceedings are cited, then the listing must contain specific addresses to which one can write to buy or borrow the reference. In general, write the paper in such a manner that its understanding does not depend on the reader having access to references that are not easily obtained.

Authors should avoid giving references to material that is posted on the internet, unless the material is truly archival, as is the case for most online journals. If referring to nonarchival material posted on the internet is necessary to give proper credit for priority, the authors should give the date at which they last viewed the material online. If authors have supplementary material that would be of interest to the readers of the article, then a proper posting of this in an archival form is to make use of the AIP Publishing's supplemental material electronic depository. Instructions for how one posts material can be found at the site <http://scitation.aip.org/content/asa/journal/jasa/info/authors>. Appropriate items for deposit include color figures, data tables, and text (e.g., appendixes) that are too lengthy or of too limited interest for inclusion in the printed journal. If authors desire to make reference to materials posted by persons other than by the authors, and if the posting is transitory, the authors should first seek to find alternate references of a more archival form that they might cite instead. In all cases, the reading of any material posted at a transitory site must not be a prerequisite to the understanding of the material in the paper itself, and when such material is cited, the authors must take care to point out that the material will not necessarily be obtainable by future readers.

In the event that a reference may be found in several places, as in the print version and the online version of a journal, refer first to the version that is most apt to be archived.

In citing articles, give both the first and last pages that include it. Including the last page will give the reader some indication of the magnitude of the article. The copying en toto of a lengthy article, for example, may be too costly for the reader's current purposes, especially if the chief objective is merely to obtain a better indication of the actual subject matter of the paper than is provided by the title.

The use of the expression *"et al."* in listing authors' names is encouraged in the body of the paper, but must not be used in the actual listing of references, as reference lists in papers are the primary sources of large data bases that persons use, among other purposes, to search by author. This rule applies regardless of the number of authors of the cited paper.

References to unpublished material in the standard format of other references must be avoided. Instead, append a graceful footnote or embed within the text a statement that you are making use of some material that you have acquired from another person—whatever material you actually use of this nature must be peripheral to the development of the principal train of thought of the paper. A critical reader will not accept its validity without at least seeing something in print. If the material is, for example, an unpublished derivation, and if the derivation is important to the substance of the present paper, then repeat the derivation in the manuscript with the original author's permission, possibly including that person as a co-author.

Journal titles must ordinarily be abbreviated, and each abbreviation must be in a "standard" form. For determination of what abbreviations to use, one can skim the reference lists that appear at the ends of recent articles in the *Journal.* The general style for making such abbreviations (e.g., Journal is always abbreviated by "J.," Applied is always abbreviated by "Appl.," International is always abbreviated by "Int.," etc.) must in any event emerge from a study of such lists, so the authors should be able to make a good guess as to the standard form. Should the guess be in error, this will often be corrected in the copy-editing process. Egregious errors are often made when the author lifts a citation from another source without actually looking up the original source. An author might be tempted, for example, to abbreviate a journal title as "Pogg. Ann.," taking this from some citation in a 19th century work. The journal cited is *Annalen der Physik,* sometimes published with the title *Annalen der Physik und Chemie,* with the standard abbreviation being "Ann. Phys. (Leipzig)." The fact that J. C. Poggendorff was at one time the editor of this journal gives very little help in the present era in distinguishing it among the astronomical number of journals that have been published. For Poggendorff's contemporaries, however, "Pogg. Ann." had a distinct meaning.

Include in references the names of publishers of book and standards and their locations. References to books and proceedings must include chapter numbers and/or page ranges.

C. Examples of reference formats

The number of possible nuances in the references that one may desire to cite is very large, and the present document cannot address all of them; a study of the reference lists at the ends of articles in recent issues in the *Journal* will resolve most questions. The following two lists, one for each of the styles mentioned above, give some representative examples for the more commonly encountered types of references. If the authors do not find a definitive applicable format in the examples below or in those they see in scanning past issues, then it is suggested that they make their best effort to create an applicable format that is consistent with the examples that they have seen, following the general principles that the information must be sufficiently complete that: (1) any present or future reader can decide whether the work is worth looking at in more detail; (2) such a reader, without great effort, can look at, borrow, photocopy, or buy a copy of the material; and (3) a citation search, based on the title, an author name, a journal name, or a publication category, will result in the present paper being matched with the cited reference.

1. Textual footnote style

[1]Y. Kawai, "Prediction of noise propagation from a depressed road by using boundary integral equations," J. Acoust. Soc. Jpn. **56**, 143–147 (2000) (in Japanese).

[2]L. S. Eisenberg, R. V. Shannon, A. S. Martinez, J. Wygonski, and A. Boothroyd, "Speech recognition with reduced spectral cues as a function of age," J. Acoust. Soc. Am. **107**, 2704–2710 (2000).

[3]J. B. Pierrehumbert, *The Phonology and Phonetics of English Intonation* (Ph.D. dissertation, Mass. Inst. Tech., Cambridge, MA, 1980); as cited by D. R. Ladd, I. Mennen, and A. Schepman, J. Acoust. Soc. Am. **107**, 2685–2696 (2000).

[4]F. A. McKiel, Jr., "Method and apparatus or sibilant classification in a speech recognition system," U.S. Patent No. 5,897,614 (27 April 1999). A brief review by D. L. Rice appears in: J. Acoust. Soc. Am. **107**, 2323 (2000).

[5]A. N. Norris, "Finite-amplitude waves in solids," in *Nonlinear Acoustics,*

edited by M. F. Hamilton and D. T. Blackstock (Academic Press, San Diego, 1998), Chap. 9, pp. 263–277.

[6]V. V. Muzychenko and S. A. Rybak, “Amplitude of resonance sound scattering by a finite cylindrical shell in a fluid,” Akust. Zh. **32**, 129–131 (1986) [Sov. Phys. Acoust. **32**, 79–80 (1986)].

[7]M. Stremel and T. Carolus, “Experimental determination of the fluctuating pressure on a rotating fan blade,” on the CD-ROM: *Berlin, March 14–19, Collected Papers, 137th Meeting of the Acoustical Society of America and the 2nd Convention of the European Acoustics Association* (ISBN 3-9804458-5-1, available from Deutsche Gesellschaft fuer Akustik, Fachbereich Physik, Universitaet Oldenburg, 26111 Oldenburg, Germany), paper 1PNSB_7.

[8]ANSI S12.60-2002 (R2009) American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools (American National Standards Institute, New York, 2002).

2. Alphabetical bibliographic list style

American National Standards Institute (**2002**). ANSI S12.60 (R2009) American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools (American National Standards Institute, New York).

Ando, Y. (**1982**). “Calculation of subjective preference in concert halls,” J. Acoust. Soc. Am. **71**(Suppl. 1), S4–S5.

Bacon, S. P. (**2000**). “Hot topics in psychological and physiological acoustics: Compression,” J. Acoust. Soc. Am. **107,** 2864(A).

Flatté, S. M., Dashen, R., Munk, W. H., Watson, K. M., and Zachariasen, F. (**1979**). *Sound Transmission through a Fluctuating Ocean* (Cambridge University Press, London), pp. 31–47.

Hamilton, W. R. (**1837**). “Third supplement to an essay on the theory of systems of waves,” Trans. Roy. Irish Soc. **17**(Pt. 1), 1–144; reprinted in: *The Mathematical Papers of Sir William Rowan Hamilton, Vol. II: Dynamics,* edited by A. W. Conway and A. J. McConnell (Cambridge University Press, London), pp. 162–211.

Helmholtz, H. (**1859**). “Theorie der Luftschwingungen in Röhren mit offenen Enden” (“Theory of air oscillations in tubes with open ends”), J. reine ang. Math. **57,** 1–72.

Kim, H.-S., Hong, J.-S., Sohn, D.-G., and Oh., J.-E. (**1999**). “Development of an active muffler system for reducing exhaust noise and flow restriction in a heavy vehicle,” Noise Control Eng. J. **47,** 57–63.

Simpson, H. J., and Houston, B. H. (**2000**). “Synthetic array measurements for waves propagating into a water-saturated sandy bottom for a smoothed and roughened surface,” J. Acoust. Soc. Am. **107,** 2329–2337.

van Bergeijk, W. A., Pierce, J. R., and David, E. E., Jr. (**1960**)**.** *Waves and the Ear* (Doubleday, Garden City, NY), Chap. 5, pp. 104–143.

Other examples may be found in the reference lists of papers recently published in the *Journal.*

D. Figure captions

The illustrations in the *Journal* have *figure captions* rather than *figure titles.* Clarity, rather than brevity, is desired, so captions can extend over several lines. Ideally, a caption must be worded so that a casual reader, on skimming an article, can obtain some indication as to what an illustration is depicting, without actually reading the text of the article. If an illustration is taken from another source, then the caption must acknowledge and cite that source. Various examples of captions can be found in the articles that appear in recent issues of the *Journal.*

If the figure will appear in black and white in the printed edition and in color online, the statement “(Color online)” should be added to the figure caption. For color figures that will appear in black and white in the printed edition of the *Journal,* the reference to colors in the figure may not be included in the caption, e.g., red circles, blue lines.

E. Acknowledgments

The section giving acknowledgments must not be numbered and must appear following the concluding section. It is preferred that acknowledgments be limited to those who helped with the research and with its formulation and to agencies and institutions that provided financial support. Administrators, administrative assistants, associate editors, and persons who assisted in the nontechnical aspects of the manuscript preparation may also be acknowledged. In many cases, sponsoring agencies require that articles give an acknowledgment and specify the format in which the acknowledgment must be stated—doing so is fully acceptable. Generally, the *Journal* expects that the page charges will be honored for any paper that carries an acknowledgment to a sponsoring organization.

F. Mathematical equations

Authors are expected to use computers with appropriate software to typeset mathematical equations.

Authors are also urged to take the nature of the actual layout of the journal pages into account when writing mathematical equations. A line in a column of text is typically 60 characters, but mathematical equations are often longer. To ensure that their papers look attractive when printed, authors must seek to write sequences of equations, each of which fits into a single column, some of which define symbols appearing in another equation, even if such results in a greater number of equations. If an equation whose length will exceed that of a single column is unavoidable, then the authors must write the equation so that it is neatly breakable into distinct segments, each of which fits into a single column. The casting of equations in a manner that requires the typesetting to revert to a single column per page (rather than two columns per page) format must be assiduously avoided. To make sure that this possibility will not occur, authors familiar with desk-top publishing software and techniques may find it convenient to temporarily recast manuscripts into a form where the column width corresponds to 60 text characters, so as to see whether none of the line breaks within equations will be awkward.

Equations are numbered consecutively in the text in the order in which they appear, the number designation is in parentheses and on the right side of the page. The numbering of the equations is independent of the section in which they appear for the main body of the text. However, for each appendix, a fresh numbering begins, so that the equations in Appendix B are labeled (B1), (B2), etc. If there is only one appendix, it is treated as if it were Appendix A in the numbering of equations.

G. Phonetic symbols

The phonetic symbols included in a JASA manuscript should be taken from the International Phonetic Alphabet (IPA), which is maintained by the International Phonetic Association, whose home page is https://www.internationalphoneticassociation.org. The display of the most recent version of the alphabet can be found at https://www.internationalphoneticassociation.org/content/full-ipa-chart.

The total set of phonetic symbols that can be used by AIP Publishing during the typesetting process is the set included among the Unicode characters. This includes most of the symbols and diacritics of the IPA chart, plus a few compiled combinations, additional tonal representations, and separated diacritics. A list of all such symbols is given in the file *phonsymbol.pdf* which can be downloaded by going to the JASA website http://scitation.aip.org/content/asa/journal/jasa/info/authors and then clicking on the item *List of Phonetic Symbols*. This file gives, for each symbol (displayed in 3 different Unicode fonts, DoulosSIL, GentiumPlus, and CharisSILCompact): its Unicode hex ID number, the Unicode character set it is part of, its Unicode character name, and its IPA definition (taken from the IPA chart). Most of these symbols and their Unicode numbers are also available from Professor John Wells of University College London at http://www.phon.ucl.ac.uk/home/wells/ipa-unicode.htm#alfa, without the Unicode character names and character set names.

The method of including such symbols in a manuscript is to use, in conjunction with a word processor, a Unicode-compliant font that includes all symbols required. Fonts that are not Unicode-compliant should not be used. Most computers come with Unicode fonts that give partial coverage of the IPA. Some sources where one can obtain Unicode fonts for Windows, MacOS, and Linux with full IPA coverage are http://www.phon.ucl.ac.uk/home/wells/ipa-unicode.htm and http://scripts.sil.org/cms/scripts/page.php?item_id=SILFontList. Further information about which fonts contain a desired symbol set can be found at http://www.alanwood.net/unicode/fontsbyrange.html#u0250 and adjacent pages at that site. While authors may use any Unicode-compliant font in their manuscript, AIP Publishing reserves the right to replace the author's font with a Unicode font of its choice (currently one of the SIL fonts Doulos, Gentium, or Charis, but subject to change in the future).

For LaTeX manuscripts, EM's LaTeX-processing environment (MikTeX) supports the use of TIPA fonts. TIPA fonts are available through the Comprehensive TeX Archive Network at http://www.ctan.org/ (download from http://www.ctan.org/pkg/tipa).

H. Figures

Each figure should be manifestly legible when reduced to one column of the printed journal page. Figures requiring the full width of a journal page are discouraged, but exceptions can be made if the reasons for such are sufficiently evident. The inclusion of figures in the manuscript should be such that the manuscript, when published, should ordinarily have no more than 30% of the space devoted to figures, and such that the total number of figures should ordinarily not be more than 20. In terms of the restriction of the total space for figures, each figure part will be considered as occupying a quarter page.

The figures are numbered in the order in which they are first referred to in the text. There must be one such referral for every figure in the text. Each figure must have a caption, and the captions are gathered together into a single list that appears at the end of the manuscript.

A chief criticism of many contemporary papers is that they contain far too many computer-generated graphical illustrations that present numerical results. An author develops a certain general computational method (realized by software) and then uses it to exhaustively discuss a large number of special cases. This practice must be avoided. Unless there is an overwhelmingly important single point that the sequence of figures demonstrates as a whole, an applicable rule of thumb is that the maximum number of figures of a given type must be four.

The clarity of most papers is greatly improved if the authors include one or more explanatory sketches. If, for example, the mathematical development presumes a certain geometrical arrangement, then a sketch of this arrangement must be included in the manuscript. If the experiment is carried out with a certain setup of instrumentation and apparatuses, then a sketch is also appropriate.

Color figures can be included in the online version of the *Journal* with no extra charge provided that these appear suitably as black and white figures in the print edition.

I. Tables

Tables are numbered by capital roman numerals (TABLE III, TABLE IV, etc.) and are collected at the end of the manuscript, following the references and preceding the figure captions, one table per page. There should be a descriptive caption (not a title) above each table in the manuscript.

Footnotes to individual items in a table are designated by raised lowercase letters (0.123[a], Martin[b], etc.) The footnotes as such are given below the table and should be as brief as practicable. If the footnotes are to references already cited in the text, then they should have forms such as— [a]Reference 10—or—[b]Firestone (1935)—depending on the citation style adopted in the text. If the reference is not cited in the text, then the footnote has the same form as a textual footnote when the alphabetical bibliographic list style is used. One would cast the footnote as in the second example above and then include a reference to a 1935 work by Firestone in the paper's overall bibliographic list. In general, it is recommended that no footnote refer to references that are not already cited in the text.

VI. THE COVER LETTER

The contents of the cover letter are usually perfunctory. There are, however, some circumstances where material in a cover letter file might be advisable or needed:

(1) If persons who would ordinarily have been included as authors have given permission or requested that their names not be included, then that must be so stated. (This requirement is imposed because some awkward situations have arisen in the past in which persons have complained that colleagues or former colleagues have deliberately omitted their names as authors from papers to which they have contributed. The *Journal* also has the policy that a paper may still be published, even if one of the persons who has contributed to the work refuses to allow his or her name to be included among the list of authors, providing there is no question of plagiarism.)

Unless a cover letter listing such exceptions is submitted, the submittal process implies that the corresponding author is attesting that the author list is complete.

(2) If there has been any prior presentation or any overlap in concept with any other manuscripts that have been either published or submitted for publication, this must be stated in a cover letter. If the manuscript has been previously submitted elsewhere for publication, and subsequently withdrawn, this must also be disclosed. If none of these apply for the submitted manuscript, then the submission process is construed to imply that the corresponding author is attesting to such a fact.

(3) (Optional.) Reasons why the authors have selected to submit their paper to JASA rather than some other journal. These would ordinarily be supplied if the authors are concerned that there may be some questions as to the paper meeting the "truly acoustics" criterion or of its being within the scope of the *Journal.* If none of the references cited in the submitted paper are to articles previously published in the *Journal,* it is highly advisable that some strong reasons be given for why the authors believe the paper falls within the scope of the *Journal.*

(4) If the online submission includes the listing of one or more persons who the authors prefer not be used as reviewers, an explanation in a cover letter would be desirable.

(5) If the authors wish to make statements which they feel are appropriate to be read by editors, but are inappropriate to be included in the actual manuscript, then such should be included in a cover letter.

Cover letters are treated by the EM system as being distinct from *rebuttal letters.*

Rebuttal letters should be submitted with revised manuscripts, and the contents are usually such that the authors give, when appropriate, rebuttals to suggestions and criticisms of the reviewers, and give detailed discussion of how and why the revised manuscript differs from what was originally submitted.

VII. EXPLANATIONS AND CATEGORIES

A. Suggestions for Associate Editors

In the suggestion of an Associate Editor who should handle a specific manuscript, authors should consult a document titled "Associate Editors identified with PACS classification items" obtainable at the JASA web site <http://scitation.aip.org/content/asa/journal/jasa/info/about>. Here the Associate Editors are identified by their initials, and the relation of the initials to the names is easily discerned from the listing of Associate Editors on the back cover of each issue, on the title page of each volume, and at the online site <http://scitation.aip.org/content/asa/journal/jasa/info/about>.

Authors are not constrained to select Associate Editors specifically identified with their choice of principal ASA Technical Committee and should note that the *Journal* has special Associate Editors for Mathematical Acoustics, Computational Acoustics, and Education in Acoustics. Review, forum, and tutorial articles are ordinarily invited; submission of unsolicited review articles, forum articles, or tutorial articles (other than those which can be construed as papers on education in acoustics) without prior discussion with the Editor-in-Chief is discouraged. Authors should suggest the Associate Editor for Education in Acoustics for tutorial papers that contain material which might be used in standard courses on acoustics or material that supplements standard textbooks.

B. Types of manuscripts

Categories of papers that are published in the *Journal* include the following:

1. Regular research articles

These are papers which report original research. There is neither a lower limit nor an upper limit on their length, although authors must pay page charges if the length results in more than 12 printed pages. The prime requirement is that such papers must contain a complete account of the reported research. These articles are classified in JASA's Table of Contents by their (most appropriate) Technical Committee or by Education in Acoustics.

2. Letters to the Editor

These are shorter research contributions that can be any of the following: (i) an announcement of a research result, preliminary to the full of the research; (ii) a scientific or technical discussion of a topic that is timely; (iii) brief alternate derivations or alternate experimental evidence concerning acoustical phenomena; (iv) provocative articles that may stimulate further research. Brevity is an essential feature of a letter, and the *Journal* suggests 3 printed journal pages as an upper limit, although it will allow up to 4 printed pages in exceptional cases.

The *Journal's* current format has been chosen so as to give letters greater prominence. Their brevity in conjunction with the possible timeliness of their contents gives impetus to a quicker processing and to a shorter time lag between submission and appearance in printed form in the *Journal.* (The quickest route to publication that the Acoustical Society currently offers is submission to the special section *JASA Express Letters* (JASA-EL) of the *Journal.* For information regarding JASA-EL, visit the site <http://scitation.aip.org/content/asa/journal/jasael/info/authors>.)

Because the desire for brevity is regarded as important, the author is not compelled to make a detailed attempt to place the work within the context of current research; the citations are relatively few and the review of related research is limited. The author should have some reason for desiring a more rapid publication than for a normal article, and the editors and the reviewers should concur with this. The work should have a modicum of completeness, to the extent that the letter "tells a story" that is at least plausible to the reader, and it should have some nontrivial support for what is being related. Not all the loose strings need be tied together. Often there is an implicit promise that the publication of the letter will be followed up by a regular research article that fills in the gaps and that does all the things that a regular research article should do.

3. Errata

These must be corrections to what actually was printed. Authors must explicitly identify the passages or equations in the paper and then state what should replace them. Long essays on why a mistake was made are not desired. A typical line in an errata article would be of the form: *Equation (23) on page 6341 is incorrect. The correct version is ... .* For detailed examples, the authors should look at previously published errata articles in the *Journal.*

4. Comments on published papers

Occasionally, one or more readers, after reading a published paper, will decide to submit a paper giving comments about that paper. The *Journal* welcomes submissions of this type, although they are reviewed to make sure that the comments are reasonable and that they are free of personal slurs. The format of the title of a comments paper is rigidly prescribed, and examples can be found in previous issues of the *Journal.* The authors of the papers under criticism are frequently consulted as reviewers, but their unsubstantiated opinion as to whether the letter is publishable is usually not given much weight.

5. Forum letters

Forum letters are analogous to the "letters to the editor" that one finds in the editorial section of major newspapers. They may express opinions or advocate actions. They may also relate anecdotes or historical facts that may be of general interest to the readers of the *Journal.* They need not have a title and should not have an abstract; they also should be brief, and they should not be of a highly technical nature. These are also submitted using the EM system, but are not handled as research articles. The applicable Associate Editor is presently the Editor-in-Chief. For examples of acceptable letters and the format that is desired, prospective authors of such letters should consult examples that have appeared in recent issues of the *Journal.*

6. Tutorial and review papers

Review and tutorial papers are occasionally accepted for publication, but are difficult to handle within the peer-review process. All are handled directly by the Editor-in-Chief, but usually with extensive discussion with the relevant Associate Editors. Usually such are invited, based on recommendations from the Associate Editors and the Technical Committees of the Society, and the tentative acceptance is based on a submitted outline and on the editors' acquaintance with the prospective author's past work. The format of such papers is similar to those of regular research articles, although there should be a table of contents following the abstract for longer research articles. Submission is handled by the online system, but the cover letter should discuss the history of prior discussions with the editors. Because of the large expenditure of time required to write an authorative review article, authors are advised not to begin writing until they have some assurance that there is a good likelihood of the submission eventually being accepted.

7. Book reviews

All book reviews must be first invited by the Associate Editor responsible for book reviews. The format for such reviews is prescribed by the Associate Editor, and the EM submittal process is used primarily to facilitate the incorporation of the reviews into the Journal.

8. Special Issues

A Special Issue must be proposed to the Editor-in-Chief by a person who is willing and able to work as a Guest Editor or coordinator, along with a regular Associate Editor. Such issues are encouraged (though not strictly required) to have an open call for papers, which will be posted on ASA's Scitation web page. Time limits for submission, review, and revision are usually enforced. If the total Special Issue is less than 100 Journal pages long, it will be printed as a part of the current Journal volume, rather than separately. Special Issues are a definite attraction for readers, and good ideas for Special Issues are always welcome.

9. Guest Invited Articles

In order to solicit papers of interest to acousticians, but outside the normal range of topics found in JASA, we have initiated the category of Guest Invited Article. These must be approved by the Editor-in-Chief, but suggestions are welcome from all ASA members.

10. Addenda

In rare cases, a small addendum may be submitted to augment a paper on a key or critical point. These are not encouraged, but can be submitted if a good case for their need is made to the Editor-in-Chief.

11. Retractions

Again, in the rare case an article has a fatal flaw, an author can contact the Editor-in-Chief about a possible retraction of the article's content. (The original article will still be part of the permanent record.)

12. Other submission categories

There are several article categories that appear on the FM submission list that are reserved for journal personnel use, and are not for general submissions. They are: Calendar, Thank You to Reviewers, Technical Committee Reports, Acoustical News, Acoustical Standards News, Reviews of Acoustical Patents, and Editorial.

VIII. FACTORS RELEVANT TO PUBLICATION DECISIONS

A. Peer review system

The *Journal* uses a peer review system in the determination of which submitted manuscripts should be published. The Associate Editors make the actual decisions; each editor has specialized understanding and prior distinguished accomplishments in the subfield of acoustics that encompasses the contributed manuscript. They seek advice from reviewers who are knowledgeable in the general subject of the paper, and the reviewers give opinions on various aspects of the work; primary questions are whether the work is original and whether it is correct. The Associate Editor and the reviewers who examine the manuscript are the authors' peers: persons with comparable standing in the same research field as the authors themselves. (Individuals interested in reviewing for JASA or for JASA-EL can convey that interest via an e-mail message to the Editor-in-Chief at <jasa@aip.org>.)

B. Selection criteria

Many submitted manuscripts are not selected for publication. Selection is based on the following factors: adherence to the stylistic requirements of the *Journal,* clarity of exposition, originality of the contribution, demonstrated understanding of previously published literature pertaining to the subject matter, appropriate discussion of the relationships of the reported research to other current research or applications, appropriateness of the subject matter to the *Journal,* correctness of the content of the article, completeness of the reporting of results, the reproducibility of the results, and the significance of the contribution. The *Journal* reserves the right to refuse publication of any submitted article without giving extensively documented reasons, although the editors usually give suggestions that can help the authors in the writing and submission of future papers. The Associate Editor also has the option, but not an obligation, of giving authors an opportunity to submit a revised manuscript addressing specific criticisms raised in the peer review process. The selection process occasionally results in mistakes, but the time limitations of the editors and the reviewers preclude extraordinary steps being taken to ensure that no mistakes are ever made. If an author feels that the decision may have been affected by an *a priori* adverse bias (such as a conflict of interest on the part of one of the reviewers), the system allows authors to express the reasons in writing and ask for an appeal review.

C. Scope of the *Journal*

Before one decides to submit a paper to the *Journal of the Acoustical Society*, it is prudent to give some thought as to whether the paper falls within the scope of the *Journal*. While this can in principle be construed very broadly, it is often the case that another journal would be a more appropriate choice. As a practical matter, the *Journal* would find it difficult to give an adequate peer review to a submitted manuscript that does not fall within the broader areas of expertise of any of its Associate Editors. In the *Journal's* peer-review process, extensive efforts are made to match a submitted manuscript with an Associate Editor knowledgeable in the field, and the Editors have the option of declining to take on the task. It is a tacit understanding that no Associate Editor should accept a paper unless he or she understands the gist of the paper and is able to make a knowledgeable assessment of the relevance of the advice of the selected reviewers. If no one wishes to handle a manuscript, the matter is referred to the Editor-in-Chief and a possible resulting decision is that the manuscript is outside the de facto scope of the *Journal.*

The *Journal* ordinarily selects for publication only articles that have a clear identification with acoustics. It would, for example, not ordinarily publish articles that report results and techniques that are not specifically applicable to acoustics, even though they could be of interest to some persons whose work is concerned with acoustics. An editorial[3] published in the October 1999 issue gives examples that are *not* clearly identifiable with acoustics.

IX. POLICIES REGARDING PRIOR PUBLICATION

The *Journal* adheres assiduously to all applicable copyright laws, and authors must not submit articles whose publication will result in a violation of such laws. Furthermore, the *Journal* follows the tradition of providing an orderly archive of scientific research in which authors take care that results and ideas are fully attributed to their originators. Conscious plagiarism is a serious breach of ethics, if not illegal. (Submission of an article that is plagiarized, in part or in full, may have serious repercussions on the future careers of the authors.) Occasionally, authors rediscover older results and submit papers reporting these results as though they were new. The desire to safeguard the *Journal* from publishing any such paper requires that submitted articles have a sufficient discussion of prior related literature to demonstrate the authors' familiarity with the literature and to establish the credibility of the assertion that the authors have carried out a thorough literature search.

In many cases, the authors themselves may have either previously circulated, published, or presented work that has substantial similarities with what is contained within the contributed manuscript. In general, JASA will not publish work that has been previously published. (An exception is when the previous publication is a letter to the editor, and when pertinent details were omitted because of the brief nature of the earlier reporting.) Presentations at conferences are not construed as prior publication; neither is the circulation of preprints or the posting of preprints on any web site, providing the site does not have the semblance of an archival online journal. Publication as such implies that the work is currently, and for the indefinite future, available, either for purchase or on loan, to a broad segment of the research community. Often the *Journal* will consider publishing manuscripts with tangible similarities to other work previously published by the authors—providing the following conditions are met: (1) the titles are different; (2) the submitted manuscript contains no extensive passages of text or figures that are the same as in the previous publication; (3) the present manuscript is a substantial update of the previous publication; (4) the previous publication has substantially less availability than would a publication in JASA; (5) the current manuscript gives ample referencing to the prior publication and explains how the current manuscript differs from the prior publication. Decisions regarding such cases are made by the Associate Editors, often in consultation with the Editor-in-Chief. (Inquiries prior to submission as to whether a given manuscript with some prior history of publication may be regarded as suitable for JASA should be addressed to the Editor-in-Chief at <jasa@aip.org>.)

The *Journal* will not consider any manuscript for publication that is presently under consideration by another journal or which is substantially similar to another one under consideration. If it should learn that such is the case, the paper will be rejected and the editors of the other journal will be notified.

Authors of an article previously published as a letter to the editor, either as a regular letter or as a letter in the JASA-EL (*JASA Express Letters*) section of the *Journal,* where the original account was either abbreviated or preliminary are encouraged to submit a more comprehensive and up-dated account of their research to the *Journal.* The same holds for POMA (Proceedings of Meetings on Acoustics) articles.

A. Speculative papers

In some cases, a paper may be largely speculative; a new theory may be offered for an as yet imperfectly understood phenomenon, without complete confirmation by experiment. Although such papers may be controversial, they often become the most important papers in the long-term development of a scientific field. They also play an important role in the stimulation of good research. Such papers are intrinsically publishable in JASA, although explicit guidelines for their selection are difficult to formulate. Of major importance are (i) that the logical development be as complete as practicable, (ii) that the principal ideas be plausible and consistent with what is currently known, (iii) that there be no known counter-examples, and (iv) that the authors give some hints as to how the ideas might be checked by future experiments or numerical computations. In addition, the authors should cite whatever prior literature exists that might indicate that others have made similar speculations.

B. Multiple submissions

The current online submittal process requires that each paper be submitted independently. Each received manuscript will be separately reviewed and judged regarding its merits for publication independently of the others. There is no formal mechanism for an author to request that two submissions, closely spaced in their times of submission, be regarded as a single submission.

In particular, the submission of two manuscripts, one labeled "Part I" and the other labeled "Part II" is not allowed. Submission of a single manuscript with the label "Part I" is also not allowed. An author may submit a separate manuscript labeled "Part II," if the text identifies which previously accepted paper is to be regarded as "Part I." Doing so may be a convenient method for alerting potential readers to the fact that the paper is a sequel to a previous paper by the author. The author should not submit a paper so labeled, however, unless the paper to be designated as "Part I" has already been accepted, either for JASA or another journal.

The Associate Editors are instructed not to process any manuscript that cannot be read without the help of as yet unpublished papers that are still under review. Consequently, authors are requested to hold back the submission of "sequels" to previously submitted papers until the disposition of those papers is determined. Alternately, authors should write the "sequels" so that the reading and comprehension of those manuscripts does not require prior reading and access of papers whose publication is still uncertain.

X. SUGGESTIONS REGARDING CONTENT

A. Introductory section

Every paper begins with introductory paragraphs. Except for short Letters to the Editor, these paragraphs appear within a separate principal section, usually with the heading "Introduction."

Although some discussion of the background of the work may be advisable, a statement of the precise subject of the work must appear within the first two paragraphs. The reader need not fully understand the subject the first time it is stated; subsequent sentences and paragraphs should clarify the statement and should supply further necessary background. The extent of the clarification must be such that a nonspecialist will be able to obtain a reasonable idea of what the paper is about. The Introduction should also explain to the nonspecialist just how the present work fits into the context of other current work done by persons other than the authors themselves. Beyond meeting these obligations, the writing should be as concise as practicable.

The Introduction must give the authors' best arguments as to why the work is original and significant. This is customarily done via a knowledgeable discussion of current and prior literature. The authors should envision typical readers or typical reviewers, and this should be a set of people that is not inordinately small, and the authors must write so as to convince them. In some cases, both originality and significance will be immediately evident to all such persons, and the arguments can be brief. In other cases, the authors may have a daunting task. It must not be assumed that readers and reviewers will give the authors the benefit of the doubt.

B. Main body of text

The writing in the main body of the paper must follow a consistent logical order. It should contain only material that pertains to the main premise of the paper, and that premise should have been stated in the Introduction. While tutorial discussions may in some places be appropriate, such should be kept to a minimum and should be only to the extent necessary to keep the envisioned readers from becoming lost.

The writing throughout the text, including the Introduction, must be in the present tense. It may be tempting to refer to subsequent sections and passages in the manuscript in the future tense, but the authors must assiduously avoid doing so, using instead phrases such as "is discussed further below."

Whenever pertinent results, primary or secondary, are reached in the progress of the paper, the writing should point out that these are pertinent results in such a manner that it would get the attention of a reader who is rapidly scanning the paper.

The requirement of a consistent logical order implies that the logical steps appear in consecutive order. Readers must not be referred to subsequent passages or to appendixes to fill in key elements of the logical development. The fact that any one such key element is lengthy or awkward is insufficient reason to relegate it to an appendix. Authors can, however, flag such passages giving the casual reader the option of skipping over them on first reading. The writing nevertheless must be directed toward the critical reader—a person who accepts no aspect of the paper on faith. (If the paper has some elements that are primarily speculative, then that should be explicitly stated, and the development should be directed toward establishing the plausibility of the speculation for the critical reader.)

To achieve clarity and readability, the authors must explicitly state the purposes of lengthy descriptions or of lengthy derivations at the beginning of the relevant passages. There should be no mysteries throughout the manuscript as to the direction in which the presentation is going.

Authors must take care that no reader becomes needlessly lost because of the use of lesser-known terminology. All terms not in standard dictionaries must be defined when they are first used. Acronyms should be avoided, but, when they are necessary, they must be explicitly defined when first used. The terminology must be consistent; different words should not be used to represent the same concept.

Efforts must be taken to avoid insulting the reader with the use of gratuitous terms or phrases such as *obvious, well-known, evident,* or *trivial.* If the adjectives are applicable, then they are unnecessary. If not, then the authors risk incurring the ill-will of the readers.

If it becomes necessary to bring in externally obtained results, then the reader must be apprised, preferably by an explicit citation to accessible literature, of the source of such results. There must be no vague allusions, such as "It has been found that..." or "It can be shown that..." If the allusion is to a mathematical derivation that the authors have themselves carried out, but which they feel is not worth describing in detail, then they should briefly outline how the derivation can be carried out, with the implication that a competent reader can fill in the necessary steps without difficulty.

For an archival journal such as JASA, reproducibility of reported results is of prime importance. Consequently, authors must give a sufficiently detailed account, so that all results, other than anecdotal, can be checked by a competent reader with comparable research facilities. If the results are numerical, then the authors must give estimates of the probable errors and state how they arrived at such estimates. (Anecdotal results are typically results of field experiments or unique case studies; such are often worth publishing as they can stimulate further work and can be used in conjunction with other results to piece together a coherent understanding of broader classes of phenomena.)

C. Concluding section

The last principal section of the article is customarily labeled "Conclusions" or "Concluding Remarks." This should not repeat the abstract, and it should not restate the subject of the paper. The wording should be directed toward a person who has some, if not thorough, familiarity with the main body of the text and who knows what the paper is all about. The authors should review the principal results of the paper and should point out just where these emerged in the body of the text. There should be a frank discussion of the limitations, if any, of the results, and there should be a broad discussion of possible implications of these results.

Often the concluding section gracefully ends with speculations on what research might be done in the future to build upon the results of the present paper. Here the authors must write in a collegial tone. There should be no remarks stating what the authors themselves intend to do next. They must be careful not to imply that the future work in the subject matter of the paper is the exclusive domain of the authors, and there should be no allusions to work in progress or to work whose publication is uncertain. It is conceivable that readers stimulated to do work along the lines suggested by the paper will contact the authors directly to avoid a duplication of effort, but that will be their choice. The spirit expressed in the paper itself should be that anyone should be free to follow-up on the suggestions made in the concluding section. A successful paper is one that does incite such interest on the part of the readers and one which is extensively cited in future papers written by persons other than the authors themselves.

D. Appendixes

The *Journal* prefers that articles not include appendixes unless there are strong reasons for their being included. Details of mathematical developments or of experimental procedures that are critical to the understanding of the substance of a paper must not be relegated to an appendix. (Authors must bear in mind that readers can easily skim over difficult passages in their first reading of a paper.) Lengthy proofs of theorems may possibly be placed in appendixes providing their stating as such in the main body of the text is manifestly plausible. Short appendixes are generally unnecessary and impede the comprehension of the paper. Appendixes may be used for lengthy tabulations of data, of explicit formulas for special cases, and of numerical results. Editors and reviewers, however, may question whether their inclusion is necessary.

E. Selection of references

References are typically cited extensively in the Introduction, and the selection of such references can play an important role in the potential usefulness of the paper to future readers and in the opinions that readers and reviewers form of the paper. No hard and fast rules can be set down as to how authors can best select references and as to how they should discuss them, but some suggestions can be found in an editorial[4] published in the May 2000 issue. If a paper falls within the scope of the *Journal,* one would ordinarily expect to find several references to papers previously published in JASA.

Demonstration of the relevance of the work is often accomplished via citations, with accompanying discussion, to recent articles in JASA and analogous journals. The implied claims to originality can be strengthened via citations, with accompanying discussion, to prior work related to the subject of the paper, sufficient to establish credibility that the authors are familiar with the literature and are not duplicating previous published work. Unsupported assertions that the authors are familiar with all applicable literature and that they have carried out an exhaustive literature survey are generally unconvincing to the critical reader.

Authors must not make large block citations of many references (e.g., four or more). There must be a stated reason for the citation of each reference, although the same reason can sometimes apply simultaneously to a small number of references. The total number of references should be kept as small a number as is consistent with the principal purposes of the paper (45 references is a suggested upper limit for a regular research article). Although nonspecialist readers may find a given paper to be informative in regard to the general state of a given field, the authors must not consciously write a research paper so that it will fulfill a dual function of being a review paper or of being a tutorial paper.

Less literate readers often form and propagate erroneous opinions concerning priority of ideas and discoveries based on the reading of recent papers, so authors must make a conscious attempt to cite original sources. Secondary sources can also be cited, if they are identified as such and especially if they are more accessible or if they provide more readable accounts. In such cases, reasons must be given as to why the secondary sources are being cited. References to individual textbooks for results that can be found in a large number of analogous textbooks should not be given, unless the cited textbook gives a uniquely clear or detailed discussion of the result. Authors should assume that any reader has access to some such textbook, and the authors should tacitly treat the result as well-known and not requiring a reference citation.

Authors must not cite any reference that the authors have not explicitly seen, unless the paper has a statement to that effect, accompanied by a statement of how the authors became aware of the reference. Such citations should be limited to crediting priority, and there must be no implied recommendations that readers should read literature which the authors themselves have not read.

F. Multimedia

A benefit of publishing in an electronic online journal is the ability to integrate multimedia files into both the published and archived articles. The online presentation of the paper allows for links to both audio and video clips directly from within the text of the article. The multimedia files submitted for *JASA* will be reviewed as part of the peer review process and accepted for publication in much the same way as are two-dimensional figures for traditional print journals. The multimedia submission guidelines presented here are subject to change because of improvements and increasing availability of the relevant technology.

The implementation of *JASA* on the Editorial Manager system is such that multimedia files are submitted in the same manner as are figure files, i.e., they are uploaded individually during the manuscript submission process. The sequence in which they are uploaded should be the same as that in which they are referred to in the text. The text should refer to these files using the designations Mm. 1, Mm. 2, etc.; this is similar to the convention of referring to figures as Fig. 1, Fig. 2, etc.

To ensure broad viewing/playing ability across hardware platforms and browsers, submissions in a variety of file formats are acceptable.

Acceptable video formats are: (i) QuickTime movies (mov); (ii) Mpeg movies (mpg); (iii) Animated Gifs (gif); (iv) Audio Video Interleave (avi).

Acceptable audio formats are: (i) AIFF (aif); (ii) Wav (wav); (iii) MP3 (mp3) at 128 kB or greater.

In the above lists, the letters in parentheses are the standard suffixes for files in the corresponding format. For example, fancymovie.mov would be a file containing a QuickTime movie.

It is important that authors make their multimedia files no larger or numerous than necessary to convey scientific information that is central to the manuscript's purpose. Authors should consider that files larger than several MB are problematic for readers using dial-up connections. Files larger than 10 MB require permission from the Editor. When video compression is used, the codec software module must be widely available. Files may not be compressed into archives, such as .zip and .tar formats. Since readers may find it tedious to download numerous files that contribute little new information, authors must select their materials carefully. Submissions with more than 6 multimedia files must receive permission from the Editor.

In the typesetting of an accepted manuscript, links will be placed within the online publication for each of the multimedia files. During the peer-review process, the reviewers and editors will access such files by going to the online site reserved for the submitted manuscript and its accompanying files, and then selecting whatever multimedia file is desired.

To help the publisher in determining just where links to each multimedia file are to be placed, authors should give a multimedia caption following the first paragraph in which the file is mentioned. The multimedia caption should resemble the following example:

Mm. 2. Fancy video file. This is a file of type "mov" (1.2 Mb).

Here "Fancy video file" is the caption for the multimedia object, which contains a level of description similar to a figure or table caption. The primary purpose of including the file type and its size is to allow readers to determine whether they wish to download it.

Authors may also wish to have a figure included for each of the video files that accompany the manuscript. One way of doing this is to take a single frame and convert it to a figure file, and then treat this in the same way as one would treat any other figure. However, the caption for such a figure should refer to the Mm number of the corresponding video file and should give a brief description of what can be found in that file.

XI. SUGGESTIONS REGARDING STYLE

A. Quality of writing and word usage

The *Journal* publishes articles in the English language only. There are very few differences of substance between British English style (as codified in the *Oxford English Dictionary*[5]) and US English style, but authors frequently must make choices in this respect, such as between alternate spelling of words that end in either *-or* or *-our,* or in either *-ized* or *-ised,* or in either *-er* or *-re.*

Articles published in JASA are expected to adhere to high standards of scholarly writing. A formal writing style free of slang is required. Good conversational skills do not necessarily translate to good formal writing skills. Authors are expected to make whatever use is necessary of standard authoritative references in regard to English grammar and writing style in preparing their manuscripts. Many good references exist—among those frequently used by professional writers are Webster's Third New International Dictionary, Unabridged,[6] Merriam-Webster's Collegiate Dictionary, 11th Edition,[7] Strunk and White's Elements of Style,[8] and the Chicago Manual of Style.[9] (The Third New International is AIP Publishing's standard dictionary.) All authors are urged to do their best to produce a high quality readable manuscript, consistent with the best traditions of scholarly and erudite writing. Occasional typographical errors and lapses of grammar can be taken care of in the copy-editing phase of the production process. Receipt of a paper whose grammatical and style errors are so excessive that they cannot be easily fixed by copy-editing will generally result in the authors being notified that the submission is not acceptable. Receipt of such a notification should not be construed as a rejection of the manuscript—the authors should take steps, possibly with external help, to revise the manuscript so that it overcomes these deficiencies. (Authors needing help or advice on scientific writing in the English language are encouraged to contact colleagues, both within and outside their own institutions, to crititque the writing in their manuscripts. Unfortunately, the staff of the *Journal* does not have the time to do this on a routine basis.)

There are some minor discrepancies in the stylistic rules that are prescribed in various references—these generally arise because of the differences in priorities that are set in different publication categories. Newspapers, for example, put high emphasis on the efficient use of limited space for conveying the news and for catching the interest of their readers. For scholarly journals, on the other hand, the overwhelming priority is clarity. In the references cited above, this is the basis for most of the stated rules. In following this tradition, the *Journal,* for example, requires a rigorous adherence to the serial comma rule (Strunk's rule number 2): In a series of three or more terms with a single conjunction, use a comma after each term except the last. Thus a JASA manuscript would refer to the "theory of Rayleigh, Helmholtz, and Kirchhoff" rather than to the "theory of Rayleigh, Helmholtz and Kirchhoff."

The priority of clarity requires that authors only use words that are likely to be understood by a large majority of potential readers. Usable words are those whose definitions may be found either in a standard unabridged English dictionary (such as the Webster's Third New International mentioned above), in a standard scientific dictionary such as the Academic Press Dictionary of Science and Technology,[10] or in a dictionary specifically devoted to acoustics such as the Dictionary of Acoustics[11] by C. L. Morfey. In some cases, words and phrases that are not in any dictionary may be *in vogue* among some workers in a given field, especially among the authors and their colleagues. Authors must give careful consideration to whether use of such terms in their manuscript is necessary; and if the authors decide to use them, precise definitions must be stated within the manuscript. Unilateral coinage of new terms by the authors is discouraged. In some cases, words with different meanings and with different spellings are pronounced exactly the same, and authors must be careful to choose the right spelling. Common errors are to interchange "principal" and "principle" and to interchange "role" and "roll."

B. Grammatical pitfalls

There are only a relatively small number of categories of errors that authors frequently make in the preparation of manuscripts. Authors should be aware of these common pitfalls and double-check that their manuscripts contain no errors in these categories. Some errors will be evident when the manuscript is read aloud; others, depending on the background of the writers, may not be. Common categories are (1) dangling participles, (2) lack of agreement in number (plural versus singular) of verbs with their subjects, (3) omission of necessary articles (such as "a," "an," and "the") that precede nouns, (4) the use of incorrect case forms (subjective, objective, possessive) for pronouns (e.g., who versus whom), and (5) use of the incorrect form (present, past, past participle, and future) in regard to tense for a verb. Individual authors may have their own peculiar pitfalls, and an independent casual reading of the manuscript by another person will generally pinpoint such pitfalls. Given the recognition that such exist, a diligent author should be able to go through the manuscript and find all instances where errors of the identified types occur.

C. Active voice and personal pronouns

Many authorities on good writing emphasize that authors should use the active rather than the passive voice. Doing so in scholarly writing, especially when mathematical expressions are present, is often infeasible, but the advice has merit. In mathematical derivations, for example, some authors use the tutorial "we" to avoid using the passive voice, so that one writes: "We substitute the expression on the right side of Eq. (5) into Eq. (2) and obtain ...," rather than: "The right side of Eq. (5) is substituted into Eq. (2), with the result being ... ." A preferable construction is to avoid the use of the tutorial "we" and to use transitive verbs such as "yields," "generates," "produces," and "leads to." Thus one would write the example above as: "Substitution of Eq. (5) into Eq. (2) yields ... ." Good writers frequently go over an early draft of a manuscript, examine each sentence and phrase written using the passive voice, and consider whether they can improve the sentence by rewriting it.

In general, personal pronouns, including the "tutorial we," are preferably avoided in scholarly writing, so that the tone is impersonal and dispassionate. In a few cases, it is appropriate that an opinion be given or that a unique personal experience be related, and personal pronouns are unavoidable. What should be assiduously avoided are any egotistical statements using personal pronouns. If a personal opinion needs to be expressed, a preferred construction is to refer to the author in the third person, such as: "the present writer believes that ... ."

D. Acronyms

Acronyms have the inconvenient feature that, should the reader be unfamiliar with them, the reader is clueless as to their meaning. Articles in scholarly journals should ideally be intelligible to many generations of future readers, and formerly common acronyms such as RCA (Radio Corporation of America, merged into the General Electric Corporation) and REA (Rural Electrification Authority) may have no meaning to such readers. Consequently, authors are requested to use acronyms sparingly and generally only when not using them would result in exceedingly awkward prose. Acronyms, such as SONAR and LASER (currently written in lowercase, sonar and laser, as ordinary words), that have become standard terms in the English language and that can be readily found in abridged dictionaries, are exceptions. If the authors use acronyms not in this category, then the meaning of the individual letters should be spelled out at the time such an acronym is first introduced.

E. Computer programs

In some cases the archival reporting of research suggests that authors give the names of specific computer programs used in the research. If the computation or data processing could just as well have been carried out with the aid of any one of a variety of such programs, then the name should be omitted. If the program has unique features that are used in the current research, then the stating of the program name must be accompanied by a brief explanation of the principal premises and functions on which the relevant features are based. One overriding consideration is that the *Journal* wishes to avoid implied endorsements of any commercial product.

F. Code words

Large research projects and large experiments that involve several research groups are frequently referred to by code words. Research articles in the *Journal* must be intelligible to a much broader group of readers, both present and future, than those individuals involved in the projects with which such a code word is associated. If possible, such code words should either not be used or else referred to in only a parenthetical sense. If attempting to do this leads to exceptionally awkward writing, then the authors must take special care to explicitly explain the nature of the project early in the paper. They must avoid any impression that the paper is specifically directed toward members of some in-group.

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[6]*Webster's Third New International Dictionary of the English Language, Unabridged,* Philip Babcock Gove, Editor-in-Chief (Merriam-Webster Inc., Springfield, MA, 1993, principal copyright 1961). This is the eighth in a series of dictionaries that has its beginning in Noah Webster's *American Dictionary of the English Language* (1828).

[7]*Merriam-Webster's Collegiate Dictionary, 11th Edition* (Merriam-Webster, Springfield, MA, 2003, principal copyright 1993). (A freshly updated version is issued annually.)

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ETHICAL PRINCIPLES OF THE ACOUSTICAL SOCIETY OF AMERICA FOR RESEARCH INVOLVING HUMAN AND NON-HUMAN ANIMALS IN RESEARCH AND PUBLISHING AND PRESENTATIONS

The Acoustical Society of America (ASA) has endorsed the following ethical principles associated with the use of human and non-human animals in research, and for publishing and presentations. The principles endorsed by the Society follow the form of those adopted by the American Psychological Association (APA), along with excerpts borrowed from the Council for International Organizations of Medical Sciences (CIOMS). The ASA acknowledges the difficulty in making ethical judgments, but the ASA wishes to set minimum socially accepted ethical standards for publishing in its journals and presenting at its meetings. These Ethical Principles are based on the principle that the individual author or presenter bears the responsibility for the ethical conduct of their research and is publication or presentation.

Authors of manuscripts submitted for publication in a journal of the Acoustical Society of America or presenting a paper at a meeting of the Society are obligated to follow the ethical principles of the Society. Failure to accept the ethical principles of the ASA shall result in the immediate rejection of manuscripts and/or proposals for publication or presentation. False indications of having followed the Ethical Principles of the ASA may be brought to the Ethics and Grievances Committee of the ASA.

APPROVAL BY APPROPRIATE GOVERNING AUTHORITY

The ASA requires all authors to abide by the principles of ethical research as a prerequisite for participation in Society-wide activities (e.g., publication of papers, presentations at meetings, etc.). Furthermore, the Society endorses the view that all research involving human and non-human vertebrate animals requires approval by the appropriate governing authority (e.g., institutional review board [IRB], or institutional animal care and use committee [IACUC], Health Insurance Portability and Accountability Act [HIPAA], or by other governing authorities used in many countries) and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, then the intent of the ASA Ethical Principles described in this document must be met. All research involving the use of human or non-human animals must have met the ASA Ethical Principles prior to the materials being submitted to the ASA for publication or presentation.

USE OF HUMAN SUBJECTS IN RESEARCH-Applicable when human subjects are used in the research

Research involving the use of human subjects should have been approved by an existing appropriate governing authority (e.g., an institutional review board [IRB]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

Informed Consent

When obtaining informed consent from prospective participants in a research protocol that has been approved by the appropriate and responsible-governing body, authors must have clearly and simply specified to the participants beforehand:

1. The purpose of the research, the expected duration of the study, and all procedures that were to be used.
2. The right of participants to decline to participate and to withdraw from the research in question after participation began.
3. The foreseeable consequences of declining or withdrawing from a study.
4. Anticipated factors that may have influenced a prospective participant's willingness to participate in a research project, such as potential risks, discomfort, or adverse effects.
5. All prospective research benefits.
6. The limits of confidentially.
7. Incentives for participation.
8. Whom to contact for questions about the research and the rights of research participants. The office/person must have willingly provided an atmosphere in which prospective participants were able to ask questions and receive answers.

Authors conducting intervention research involving the use of experimental treatments must have clarified, for each prospective participant, the following issues at the outset of the research:

1. The experimental nature of the treatment;
2. The services that were or were not to be available to the control group(s) if appropriate;
3. The means by which assignment to treatment and control groups were made;
4. Available treatment alternatives if an individual did not wish to participate in the research or wished to withdraw once a study had begun; and
5. Compensation for expenses incurred as a result of participating in a study including, if appropriate, whether reimbursement from the participant or a third-party payer was sought.

Informed Consent for Recording Voices and Images in Research

Authors must have obtained informed consent from research participants prior to recording their voices or images for data collection unless:

1. The research consisted solely of naturalistic observations in public places, and it was not anticipated that the recording would be used in a manner that could have caused personal identification or harm, or
2. The research design included deception. If deceptive tactics were a necessary component of the research design, consent for the use of recordings was obtained during the debriefing session.

Client/Patient, Student, and Subordinate Research Participants

When authors conduct research with clients/patients, students, or subordinates as participants, they must have taken steps to protect the prospective participants from adverse consequences of declining or withdrawing from participation.

Dispensing With Informed Consent for Research

Authors may have dispensed with the requirement to obtain informed consent when:

1. It was reasonable to assume that the research protocol in question did not create distress or harm to the participant and involves:
    a. The study of normal educational practices, curricula, or classroom management methods that were conducted in educational settings
    b. Anonymous questionnaires, naturalistic observations, or archival research for which disclosure of responses would not place participants at risk of criminal or civil liability or damage their financial standing, employability, or reputation, and confidentiality
    c. The study of factors related to job or organization effectiveness conducted in organizational settings for which there was no risk to participants' employability, and confidentiality.
2. Dispensation is permitted by law.
3. The research involved the collection or study of existing data, documents, records, pathological specimens, or diagnostic specimens, if these sources are publicly available or if the information is recorded by the investigator in such a manner that subjects cannot be identified, directly or through identifiers linked to the subjects.

Offering Inducements for Research Participation

(a) Authors must not have made excessive or inappropriate financial or other inducements for research participation when such inducements are likely to coerce participation.

(b) When offering professional services as an inducement for research participation, authors must have clarified the nature of the services, as well as the risks, obligations, and limitations.

Deception in Research

(a) Authors must not have conducted a study involving deception unless they had determined that the use of deceptive techniques was justified by the study's significant prospective scientific, educational, or applied value and that effective non-deceptive alternative procedures were not feasible.

(b) Authors must not have deceived prospective participants about research that is reasonably expected to cause physical pain or severe emotional distress.

(c) Authors must have explained any deception that was an integral feature of the design and conduct of an experiment to participants as early as was feasible, preferably at the conclusion of their participation, but no later than at the conclusion of the data collection period, and participants were freely permitted to withdraw their data.

Debriefing

(a) Authors must have provided a prompt opportunity for participants to obtain appropriate information about the nature, results, and conclusions of the research project for which they were a part, and they must have taken reasonable steps to correct any misconceptions that participants may have had of which the experimenters were aware.

(b) If scientific or humane values justified delaying or withholding relevant information, authors must have taken reasonable measures to reduce the risk of harm.

(c) If authors were aware that research procedures had harmed a participant, they must have taken reasonable steps to have minimized the harm.

HUMANE CARE AND USE OF NON-HUMAN VERTEBRATE ANIMALS IN RESEARCH-Applicable when non-human vertebrate animals are used in the research

The advancement of science and the development of improved means to protect the health and well being both of human and non-human vertebrate animals often require the use of intact individuals representing a wide variety of species in experiments designed to address reasonable scientific questions. Vertebrate animal experiments should have been undertaken only after due consideration of the relevance for health, conservation, and the advancement of scientific knowledge. (Modified from the Council for International Organizations of Medical Science (CIOMS) document: "International Guiding Principles for Biomedical Research Involving Animals1985"). Research involving the use of vertebrate animals should have been approved by an existing appropriate governing authority (e.g., an institutional animal care and use committee [IACUC]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

The proper and humane treatment of vertebrate animals in research demands that investigators:

1. Acquired, cared for, used, interacted with, observed, and disposed of animals in compliance with all current federal, state, and local laws and regulations, and with professional standards.
2. Are knowledgeable of applicable research methods and are experienced in the care of laboratory animals, supervised all procedures involving animals, and assumed responsibility for the comfort, health, and humane treatment of experimental animals under all circumstances.
3. Have insured that the current research is not repetitive of previously published work.
4. Should have used alternatives (e.g., mathematical models, computer simulations, etc.) when possible and reasonable.
5. Must have performed surgical procedures that were under appropriate anesthesia and followed techniques that avoided infection and minimized pain during and after surgery.
6. Have ensured that all subordinates who use animals as a part of their employment or education received instruction in research methods and in the care, maintenance, and handling of the species that were used, commensurate with the nature of their role as a member of the research team.
7. Must have made all reasonable efforts to minimize the number of vertebrate animals used, the discomfort, the illness, and the pain of all animal subjects.
8. Must have made all reasonable efforts to minimize any harm to the environment necessary for the safety and well being of animals that were observed or may have been affective as part of a research study.
9. Must have made all reasonable efforts to have monitored and then mitigated any possible adverse affects to animals that were observed as a function of the experimental protocol.
10. Who have used a procedure subjecting animals to pain, stress, or privation may have done so only when an alternative procedure was unavailable; the goal was justified by its prospective scientific, educational, or applied value; and the protocol had been approved by an appropriate review board.
11. Proceeded rapidly to humanely terminate an animal's life when it was necessary and appropriate, always minimizing pain and always in accordance with accepted procedures as determined by an appropriate review board.

PUBLICATION and PRESENTATION ETHICS-For publications in ASA journals and presentations at ASA sponsored meetings

Plagiarism

Authors must not have presented portions of another's work or data as their own under any circumstances.

Publication Credit

Authors have taken responsibility and credit, including authorship credit, only for work they have actually performed or to which they have substantially contributed. Principal authorship and other publication credits accurately reflect the relative scientific or professional contributions of the individuals involved, regardless of their relative status. Mere possession of an institutional position, such as a department chair, does not justify authorship credit. Minor contributions to the research or to the writing of the paper should have been acknowledged appropriately, such as in footnotes or in an introductory statement.

Duplicate Publication of Data

Authors did not publish, as original data, findings that have been previously published. This does not preclude the republication of data when they are accompanied by proper acknowledgment as defined by the publication policies of the ASA.

Reporting Research Results

If authors discover significant errors in published data, reasonable steps must be made in as timely a manner as possible to rectify such errors. Errors can be rectified by a correction, retraction, erratum, or other appropriate publication means.

DISCLOSURE OF CONFLICTS OF INTEREST

If the publication or presentation of the work could directly benefit the author(s), especially financially, then the author(s) must disclose the nature of the conflict:

1) The complete affiliation(s) of each author and sources of funding for the published or presented research should be clearly described in the paper or publication abstract.

2) If the publication or presentation of the research would directly lead to the financial gain of the authors(s), then a statement to this effect must appear in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper.

3) If the research that is to be published or presented is in a controversial area and the publication or presentation presents only one view in regard to the controversy, then the existence of the controversy and this view must be provided in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper. It is the responsibility of the author to determine if the paper or presentation is in a controversial area and if the person is expressing a singular view regarding the controversy.

Sustaining Members of the Acoustical Society of America

The Acoustical Society is grateful for the financial assistance being given by the Sustaining Members listed below and invites applications for sustaining membership from other individuals or corporations who are interested in the welfare of the Society.

Application for membership may be made to the Executive Director of the Society and is subject to the approval of the Executive Council. Dues of $1000.00 for small businesses (annual gross below $100 million) and $2000.00 for large businesses (annual gross above $100 million or staff of commensurate size) include a subscription to the *Journal* as well as a yearly membership certificate suitable for framing. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year.

Additional information and application forms may be obtained from Elaine Moran, Office Manager, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: elaine@acousticalsociety.org

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Measurement microphones, Intensity probes, Calibrators

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Yorktown Heights, New York
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Knowles Electronics, Inc.
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Massa Products Corporation
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Manufacture Loudspeakers and Acoustical Test Equipment

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www.ncac.com
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An Association of Independent Firms Consulting in Acoustics

Raytheon Company
Integrated Defense Systems
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Portsmouth, Rhode Island
Sonar Systems and Oceanographic Instrumentation: R&D in Underwater Sound Propagation and Signal Processing

ROXUL, Inc. – Core Solutions (OEM)
www.roxul.com
Milton, ON, Canada
Offers a variety of insulation products ranging in density and dimension to meet any production requirements. Products are successfully used in numerous acoustical OEM applications providing solutions for a number of industries

Shure Incorporated
www.shure.com
Niles, Illinois
Design, development, and manufacture of cabled and wireless microphones for broadcasting, professional recording, sound reinforcement, mobile communications, and voice input–output applications; audio circuitry equipment; high fidelity phonograph cartridges and styli: automatic mixing systems; and related audio components and accessories. The firm was founded in 1925.

Thales Underwater Systems
www.thales-naval.com
Somerset, United Kingdom
Prime contract management, customer support services, sonar design and production, masts and communications systems design and production

3M Occupational Health & Environmental Safety Division
www.3m.com/occsafety
Minneapolis, Minnesota
Products for personal and environmental safety, featuring E·A·R and Peltor brand hearing protection and fit testing, Quest measurement instrumentation, audiological devices, materials for control of noise, vibration, and mechanical energy, and the E·A·RCALSM laboratory for research, development, and education, NVLAP-accredited since 1992.
Hearing conservation resource center
www.e-a-r.com/hearingconservation

Wenger Corporation
www.wengercorp.com
Owatonna, Minnesota
Design and Manufacturing of Architectural
Acoustical Products including Absorbers, Diffusers, Modular Sound Isolating Practice Rooms, Acoustical Shells and Clouds for Music Rehearsal and Performance Spaces

Wyle Laboratories
www.wyle.com
Arlington, Virginia
The Wyle Acoustics Group provides a wide range of professional services focused on acoustics, vibration, and their allied technologies, including services to the aviation industry

ACOUSTICAL · SOCIETY · OF · AMERICA

APPLICATION FOR SUSTAINING MEMBERSHIP

The Bylaws provide that any person, corporation, or organization contributing annual dues as fixed by the Executive Council shall be eligible for election to Sustaining Membership in the Society.

Dues have been fixed by the Executive Council as follows: $1000 for small businesses (annual gross below $100 million); $2000 for large businesses (annual gross above $100 million or staff of commensurate size). Dues include one year subscription to *The Journal of the Acoustical Society of America* and programs of Meetings of the Society. Please do not send dues with application. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year. If elected, you will be billed.

Name of Company ____________________

Address ____________________

Telephone: ____________________ Fax: ____________________

E-mail: ____________________ WWW: ____________________

Size of Business: ☐ Small business ☐ Small business—No Journal ☐ Large business

Type of Business ____________________

Please enclose a copy of your organization's brochure.

In listing of Sustaining Members in the Journal and on the ASA homepage we should like to indicate our products or services as follows:

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(please do not exceed fifty characters)

Name of company representative to whom journal should be sent:

____

It is understood that a Sustaining Member will not use the membership for promotional purposes.

Signature of company representatives making application:

____

Please send completed applications to: Executive Director, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, (516) 576-2360, asa@acousticalsociety.org

MEMBERSHIP INFORMATION AND APPLICATION INSTRUCTIONS

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of the application; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

BENEFITS OF MEMBERSHIP	Full Member	Associate	ce-Associate	Student
JASA Online–Vol. 1 (1929) to present	*	*	*	*
JASA tables of contents e-mail alerts	*	*	*	*
JASA, printed or CD ROM	*	*		
JASA Express Letters–online	*	*	*	*
Acoustics Today–the quarterly magazine	*	*	*	*
Proceedings of Meetings on Acoustics	*	*	*	*
Noise Control and *Sound*, *It's Uses and Control*–online archival magazines	*	*	*	*
Acoustics Research Letters Online (ARLO)–online archive	*	*	*	*
Programs for Meetings	Online	Online	Online	Online
Meeting Calls for Papers	Online	Online	Online	Online
Reduced Meeting Registration Fees	*	*		*
5 Free ASA standards per year-download only	*	*		*
Standards Discounts	*	*		*
Society Membership Directory	Online	Online	Online	Online
Electronic Announcements	*	*	*	*
Physics Today	*	*	*	*
Eligibility to vote and hold office in ASA	*			
Eligibility to be elected Fellow	*	*		
Participation in ASA Committees	*	*	*	*

QUALIFICATIONS FOR EACH GRADE OF MEMBERSHIP AND ANNUAL DUES

Student: Any student interested in acoustics who is enrolled in an accredited college or university for half time or more (at least eight semester hours). Dues: $45 per year.
Associate: Any individual interested in acoustics. Dues: $95 per year. After five years, the dues of an Associate increase to that of a full Member.
Corresponding Electronic Associate: Any individual residing in a developing country who wishes to have access to ASA's online publications only including *The Journal of the Acoustical Society of America* and Meeting Programs [see http://acousticalsociety.org/membership/membership_and_benefits]. Dues $45 per year.
Member: Any person active in acoustics, who has an academic degree in acoustics or in a closely related field or who has had the equivalent of an academic degree in scientific or professional experience in acoustics, shall be eligible for election to Membership in the Society. A nonmember applying for full Membership will automatically be made an interim Associate Member, and must submit $95 with the application for the first year's dues. Election to full Membership may require six months or more for processing; dues as a full Member will be billed for subsequent years.

JOURNAL OPTIONS AND COSTS FOR FULL MEMBERS AND ASSOCIATE MEMBERS ONLY

- ONLINE JOURNAL. All members will receive access to the *The Journal of the Acoustical Society of America (JASA)* at no charge in addition to dues.
- PRINT JOURNAL. Twelve monthly issues of *The Journal of the Acoustical Society of America*. **Cost: $35 in addition to dues.**
- CD-ROM. The CD ROM mailed bimonthly. This option includes all of the material published in the Journal on CD ROM. **Cost: $35 in addition to dues.**
- COMBINATION OF THE CD-ROM AND PRINTED JOURNAL. The CD-ROM mailed bimonthly and the printed journal mailed monthly. **Cost: $70 in addition to dues.**
- **EFFECTIVE DATE OF MEMBERSHIP**. If your application for membership and dues payment are received by 15 September, your membership and Journal subscription will begin during the current year and you will receive all back issues for the year. If you select the print journal option. If your application is received after 15 September, however, your dues payment will be applied to the following year and your Journal subscription will begin the following year.

OVERSEAS AIR DELIVERY OF JOURNALS

Members outside North, South, and Central America can choose to have print journals sent by air freight at a cost of $165 in addition to dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

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APPLICATION FOR MEMBERSHIP

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of this form; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

PART I. TO BE COMPLETED BY ALL APPLICANTS (Please print or type all entries)

CHECK ONE BOX IN EACH COLUMN ON THE RIGHT	☐ NON-MEMBER APPLYING FOR: ☐ MEMBER REQUESTING TRANSFER TO:	☐ STUDENT MEMBERSHIP ☐ ASSOCIATE MEMBERSHIP ☐ CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP ☐ FULL MEMBERSHIP	Note that your choice of journal option *may* increase or decrease the amount you must remit.

SELECT JOURNAL OPTION:

Student members will automatically receive access to The Journal of the Acoustical Society of America online at no charge in addition to dues. Remit $45. (Note: Student members may also receive the Journal on CD ROM at an additional charge of $35.)

Corresponding Electronic Associate Members will automatically receive access to The Journal of the Acoustical Society of America and Meeting Programs online at no charge in addition to dues. Remit $45.

Applicants for **Associate or full Membership** must select one Journal option from those listed below. Note that your selection of journal option determines the amount you must remit.

[ ] Online access only—$95
[ ] Online access plus print Journal $130
[ ] Online access plus CD ROM—$130
[ ] Online access plus print Journal and CD ROM combination—$165

Applications received after 15 September: Membership and Journal subscriptions begin the following year.

OPTIONAL AIR DELIVERY: Applicants from outside North, South, and Central America may choose air freight delivery of print journals for an additional charge of $165. If you wish to receive journals by air, remit the additional amount owed with your dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

LAST NAME	FIRST NAME	MIDDLE INITIAL	MS/MR/MRS/DR/PROF
HOME ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
NAME OF ORGANIZATION OR BUSINESS			
DEPARTMENT			
ORGANIZATION ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
BUSINESS TELEPHONE: AREA CODE/NUMBER	MOBILE PHONE: AREA CODE/NUMBER	HOME TELEPHONE: AREA CODE/NUMBER	
E-MAIL ADDRESS: (PRINT CLEARLY)			
DATE AND PLACE OF BIRTH (Req'd for Awards and Emeritus Status)	SEX: ☐ Female ☐ Male		
HIGHEST ACADEMIC DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE
OTHER DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE

CHECK PERFERRED ADDRESS FOR MAIL: ☐ HOME ☐ ORGANIZATION

Part I Continued ➔

PART I CONTINUED: ACOUSTICAL AREAS OF INTEREST TO APPLICANT. Indicate your three main areas of interest below, using 1 for your main interest, 2 for your second, and 3 for your third interest. (DO NOT USE CHECK MARKS.)

- ☐ ACOUSTICAL OCEANOGRAPHY **M**
- ☐ ANIMAL BIOACOUSTICS **L**
- ☐ ARCHITECTURAL ACOUSTICS **A**
- ☐ BIOMEDICAL ACOUSTICS **K**
- ☐ ENGINEERING ACOUSTICS **B**
- ☐ MUSICAL ACOUSTICS **C**
- ☐ NOISE & NOISE CONTROL **D**
- ☐ PHYSICAL ACOUSTICS **E**
- ☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
- ☐ SIGNAL PROCESSING IN ACOUSTICS **N**
- ☐ SPEECH COMMUNICATION **H**
- ☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
- ☐ UNDERWATER ACOUSTICS **J**

PART II: APPLICATION FOR STUDENT MEMBERSHIP

NAME AND ADDRESS OF COLLEGE OR UNIVERSITY WHERE PRESENTLY ENROLLED		
DEGREE EXPECTED	MONTH & YEAR DEGREE EXPECTED	NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER
PRINT NAMES & E-MAIL ADDRESSES OF TWO FACULTY MEMBERS CERTIFYING THAT YOU ARE REGISTERED FOR AT LEAST ONE-HALF OF FULL TIME		
SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME		
SIGNATURE OF APPLICANT		DATE

PART III: APPLICATION FOR ASSOCIATE MEMBERSHIP, CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP OR FULL MEMBERSHIP (and interim Associate Membership)

SUMMARIZE YOUR MAJOR PROFESSIONAL EXPERIENCE on the lines below: list employers, duties and position titles, and dates, beginning with your present position. Attach additional sheets if more space is required.

CONTRIBUTIONS TO ACOUSTICS: LIST MAIN PUBLICATIONS, PATENTS, ETC. Attach separate sheets if required.

SPONSORS AND REFERENCES: An application for full Membership requires the names, addresses, and signatures of two references who must be **full Members or Fellows** of the Acoustical Society. Names and signatures are NOT required for Associate Membership, Corresponding Electronic Associate Membership or Student Membership applications.

PRINT NAME OF REFERENCE (required for Full Member applications only)	PRINT NAME OF SECOND REFERENCE (required for Full Member applications only)
ADDRESS OF REFERENCE	ADDRESS OF SECOND REFERENCE
SIGNATURE OF REFERENCE (required for Full Member applications only)	SIGNATURE OF SECOND REFERENCE (required for Full Member applications only)
SIGNATURE OF APPLICANT	DATE

MAIL THIS COMPLETED APPLICATION, WITH APPROPRIATE PAYMENT TO: ACOUSTICAL SOCIETY OF AMERICA, 1305 WALT WHITMAN ROAD, SUITE 300, MELVILLE, NY 11747-4300.

METHOD OF PAYMENT

☐ Check or money order enclosed for $ ____________ (U.S. funds/drawn on U.S. bank)

☐ American Express ☐ VISA ☐ MasterCard Signature ________________________________

(Credit card orders must be signed)

Account Number ☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐

Expiration Date Mo. ☐☐ Yr. ☐☐

Security Code ☐☐☐☐

Due to security risks and Payment Card Industry (PCI) data security standards e-mail is NOT an acceptable way to transmit credit card information. Please return this form by Fax (631-923-2875) or by postal mail.

Regional Chapters and Student Chapters

Anyone interested in becoming a member of a regional chapter or in learning if a meeting of the chapter will be held while he/she is in the local area of the chapter, either permanently or on travel, is welcome to contact the appropriate chapter representative. Contact information is listed below for each chapter representative.

Anyone interested in organizing a regional chapter in an area not covered by any of the chapters below is invited to contact the Cochairs of the Committee on Regional Chapters for information and assistance: Evelyn M. Hoglund, Ohio State University, Columbus, OH 43204, hoglund1@osu.edu and Sandra Guzman, Columbia College Chicago, Chicago, IL 60605, sguzman@colum.edu

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ACOUSTICAL DESIGN OF MUSIC EDUCATION FACILITIES. Edward R. McCue and Richard H. Talaske, Eds. Plans, photographs, and descriptions of 50 facilities with explanatory text and essays on the design process. 236 pp, paper, 1990. Price: $23. **Item # 0-88318-8104**

ACOUSTICAL DESIGN OF THEATERS FOR DRAMA PERFORMANCE: 1985–2010. David T. Bradley, Erica E. Ryherd, & Michelle C. Vigeant, Eds. Descriptions, color images, and technical and acoustical data of 130 drama theatres from around the world, with an acoustics overview, glossary, and essays reflecting on the theatre design process. 334 pp, hardcover 2010. Price: $45. **Item # 978-0-9846084-5-4**

ACOUSTICAL DESIGNING IN ARCHITECTURE. Vern O. Knudsen and Cyril M. Harris. Comprehensive, non-mathematical treatment of architectural acoustics; general principles of acoustical designing. 408 pp, paper, 1980 (original published 1950). Price: $23. **Item # 0-88318-267X**

ACOUSTICAL MEASUREMENTS. Leo L. Beranek. Classic text with more than half revised or rewritten. 841 pp, hardcover 1989 (original published 1948). **Available on Amazon.com**

ACOUSTICS. Leo L. Beranek. Source of practical acoustical concepts and theory, with information on microphones, loudspeakers and speaker enclosures, and room acoustics. 491 pp, hardcover 1986 (original published 1954). **Available on Amazon.com**

ACOUSTICS—AN INTRODUCTION TO ITS PHYSICAL PRINCIPLES AND APPLICATIONS. Allan D. Pierce. Textbook introducing the physical principles and theoretical basis of acoustics, concentrating on concepts and points of view that have proven useful in applications such as noise control, underwater sound, architectural acoustics, audio engineering, nondestructive testing, remote sensing, and medical ultrasonics. Includes problems and answers. 678 pp, hardcover 1989 (original published 1981). Price: $33. **Item # 0-88318-6128**

ACOUSTICS, ELASTICITY AND THERMODYNAMICS OF POROUS MEDIA: TWENTY-ONE PAPERS BY M. A. BIOT. Ivan Tolstoy, Ed. Presents Biot's theory of porous media with applications to acoustic wave propagation, geophysics, seismology, soil mechanics, strength of porous materials, and viscoelasticity. 272 pp, hardcover 1991. Price: $28. **Item # 1-56396-0141**

ACOUSTICS OF AUDITORIUMS IN PUBLIC BUILDINGS. Leonid I. Makrinenko, John S. Bradley, Ed. Presents developments resulting from studies of building physics. 172 pp, hardcover 1994 (original published 1986). Price: $38. **Item # 1-56396-3604**

ACOUSTICS OF WORSHIP SPACES. David Lubman and Ewart A. Wetherill, Eds. Drawings, photographs, and accompanying data of worship houses provide information on the acoustical design of chapels, churches, mosques, temples, and synagogues. 91 pp, paper 1985. Price: $23. **OUT-OF-PRINT**

ASA EDITION OF SPEECH AND HEARING IN COMMUNICATION. Harvey Fletcher; Jont B. Allen, Ed. A summary of Harvey Fletcher's 33 years of acoustics work at Bell Labs. A new introduction, index, and complete bibliography of Fletcher's work are important additions to this classic volume. 487 pp, hardcover 1995 (original published 1953). Price: $40. **Item # 1-56396-3930**

AEROACOUSTICS OF FLIGHT VEHICLES: THEORY AND PRACTICE. Harvey H. Hubbard, Ed. Two-volumes oriented toward flight vehicles emphasizing the underlying concepts of noise generation, propagation, predicting and control. Vol. 1 589 pp/Vol. 2 426 pp, hardcover 1994 (original published 1991). Price per 2-vol. set: $58. **Item # 1-56396-404X**

COLLECTED PAPERS ON ACOUSTICS. Wallace Clement Sabine. Classic work on acoustics for architects and acousticians. 304 pp, hardcover 1993 (originally published 1921). Price: $28. **Item # 0-932146-600**

CONCERT HALLS AND OPERA HOUSES. Leo L. Beranek. Over 200 photos and architectural drawings of 100 concert halls and opera houses in 31 countries with rank-ordering of 79 halls and houses according to acoustical quality. 653 pp. hardcover 2003. Price: $50. **OUT-OF-PRINT**

CRYSTAL ACOUSTICS. M.J.P. Musgrave. For physicists and engineers who study stress-wave propagation in anisotropic media and crystals. 406 pp. hardcover (originally published 1970). Price: $34. **Item # 0-9744067-0-8**

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THE EAR AS A COMMUNICATION RECEIVER. Eberhard Zwicker & Richard Feldtkeller. Translated by Hannes Müsch, Søren Buus, Mary Florentine. Translation of the classic *Das Ohr Als Nachrichtenempfänger*. Aimed at communication engineers and sensory psychologists. Comprehensive coverage of the excitation pattern model and loudness calculation schemes. 297 pp, hardcover 1999 (original published 1967). Price: $50. **OUT-OF-PRINT**

ELECTROACOUSTICS: THE ANALYSIS OF TRANSDUCTION, AND ITS HISTORICAL BACKGROUND. Frederick V. Hunt. Analysis of the conceptual development of electroacoustics including origins of echo ranging, the crystal oscillator, evolution of the dynamic loudspeaker, and electromechanical coupling, 260 pp, paper 1982 (original published 1954). **Available on Amazon.com**

ELEMENTS OF ACOUSTICS. Samuel Temkin. Treatment of acoustics as a branch of fluid mechanics. Main topics include propagation in uniform fluids at rest, trans-mission and reflection phenomena, attenuation and dispersion, and emission. 515 pp. hardcover 2001 (original published 1981). Price: $30. **Item # 1-56396-997-1**

EXPERIMENTS IN HEARING. Georg von Békésy. Classic on hearing containing vital roots of contemporary auditory knowledge. 760 pp, paper 1989 (original published 1960). Price: $23. **Item # 0-88318-6306**

FOUNDATIONS OF ACOUSTICS. Eugen Skudrzyk. An advanced treatment of the mathematical and physical foundations of acoustics. Topics include integral transforms and Fourier analysis, signal processing, probability and statistics, solutions to the wave equation, radiation and diffraction of sound. 790 pp. hardcover 2008 (originally published 1971). Price: $60. **Item # 3-211-80988-0**

HALLS FOR MUSIC PERFORMANCE: TWO DECADES OF EXPERIENCE, 1962–1982. Richard H. Talaske, Ewart A. Wetherill, and William J. Cavanaugh, Eds. Drawings, photos, and technical and physical data on 80 halls; examines standards of quality and technical capabilities of performing arts facilities. 192 pp, paper 1982. Price: $23. **OUT-OF-PRINT**

HALLS FOR MUSIC PERFORMANCE: ANOTHER TWO DECADES OF EXPERIENCE 1982–2002. Ian Hoffman, Christopher Storch, and Timothy Foulkes, Eds. Drawings, color photos, technical and physical data on 142 halls. 301 pp, hardcover 2003. Price: $56. **Item # 0-9744067-2-4**

HANDBOOK OF ACOUSTICAL MEASUREMENTS AND NOISE CONTROL, THIRD EDITION. Cyril M. Harris. Comprehensive coverage of noise control and measuring instruments containing over 50 chapters written by top experts in the field. 1024 pp, hardcover 1998 (original published 1991). **OUT-OF-PRINT**

HEARING: ITS PSYCHOLOGY AND PHYSIOLOGY. Stanley Smith Stevens & Hallowell Davis. Volume leads readers from the fundamentals of the psycho-physiology of hearing to a complete understanding of the anatomy and physiology of the ear. 512 pp, paper 1983 (originally published 1938). **OUT-OF-PRINT**

NONLINEAR ACOUSTICS. Mark F. Hamilton and David T. Blackstock. Research monograph and reference for scientists and engineers, and textbook for a graduate course in nonlinear acoustics. 15 chapters written by leading experts in the field. 455 pp, hardcover, 2008 (originally published in 1996). Price: $45. **Item # 0-97440-6759**

NONLINEAR ACOUSTICS. Robert T. Beyer. A concise overview of the depth and breadth of nonlinear acoustics with an appendix containing references to new developments. 452 pp, hardcover 1997 (originally published 1974). Price: $45. **Item # 1-56396-724-3**

NONLINEAR UNDERWATER ACOUSTICS. B. K. Novikov, O. V. Rudenko, V. I. Timoshenko. Translated by Robert T. Beyer. Applies the basic theory of nonlinear acoustic propagation to directional sound sources and receivers, including design nomographs and construction details of parametric arrays. 272 pp., paper 1987. Price: $34. **Item # 0-88318-5229**

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