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168th Meeting
Acoustical Society of America

Indianapolis Marriott Downtown Hotel
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The Journal of the Acoustical Society of America

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TECHNICAL PROGRAM SUMMARY
***Indicates Special Session**

Monday morning

1aAB Topics in Animal Bioacoustics I
*1aNS Metamaterials for Noise Control I
*1aPA Jet Noise Measurements and Analyses I
1aSC Speech Processing and Technology (Poster Session)
*1aSP Sampling Methods for Bayesian Signal Processing
*1aUW Understanding the Target/Waveguide System–Measurement and Modeling I

Monday afternoon

*1pAA Computer Auralization as an Aid to Acoustically Proper Owner/Architect Design Decisions
*1pAB Array Localization of Vocalizing Animals
1pBA Medical Ultrasound
*1pNS Metamaterials for Noise Control II
*1pPA Jet Noise Measurements and Analyses II
*1pSCa Findings and Methods in Ultrasound Speech Articulation Tracking
1pSCb Issues in Cross Language and Dialect Perception (Poster Session)
*1pUW Understanding the Target/Waveguide System–Measurement and Modeling II

Monday evening

*1eID Tutorial Lecture on Musical Acoustics: Science and Performance

Tuesday morning

*2aAA Architectural Acoustics and Audio I
*2aAB Mobile Autonomous Platforms for Bioacoustic Sensing
*2aAO Parameter Estimation in Environments That Include Out-of-Plane Propagation Effects
*2aBA Quantitative Ultrasound I
*2aED Undergraduate Research Exposition (Poster Session)
*2aID Historical Transducers
*2aMU Piano Acoustics
*2aNSa New Frontiers in Hearing Protection I
*2aNSb Launch Vehicle Acoustics I
2aPA Outdoor Sound Propagation
*2aSAa Computational Methods in Structural Acoustics and Vibration
*2aSAb Vehicle Interior Noise
2aSC Speech Production and Articulation (Poster Session)
2aUW Signal Processing and Ambient Noise

Tuesday afternoon

*2pAA Architectural Acoustics and Audio II
2pAB Topics in Animal Bioacoustics II
2pAO General Topics in Acoustical Oceanography
*2pBA Quantitative Ultrasound II
2pEDa General Topics in Education in Acoustics
*2pEDb Take's 5
*2pID Centennial Tribute to Leo Beranek's Contributions in Acoustics
*2pMU Synchronization Models in Musical Acoustics and Psychology
*2pNSa New Frontiers in Hearing Protection II
*2pNSb Launch Vehicle Acoustics II
*2pPA Demonstrations in Acoustics
*2pSA Nearfield Acoustical Holography
2pSC Segments and Suprasegmentals (Poster Session)
2pUW Propagation and Scattering

Wednesday morning

*3aAA Design and Performance of Office Workspaces in High Performance Buildings
*3aAB Predator–Prey Relationships
*3aAO Education in Acoustical Oceanography and Underwater Acoustics
3aBA Kidney Stone Lithotripsy
*3aEA Mechanics of Continuous Media
*3aID Graduate Studies in Acoustics (Poster Session)
3aMU Topics in Musical Acoustics
*3aNS Wind Turbine Noise
*3aPA Acoustics of Pile Driving: Models, Measurements, and Mitigation
*3aSAa Vibration Reduction in Air-Handling Systems
3aSAb General Topics in Structural Acoustics and Vibration
*3aSC Vowels = Space + Time, and Beyond: A Session in Honor of Diane Kewley-Port
3aSPa Beamforming and Source Tracking
3aSPb Spectral Analysis, Source Tracking, and System Identification (Poster Session)
*3aUW Standardization of Measurement, Modeling, and Terminology of Underwater Sound

Wednesday afternoon

3pAA Architectural Acoustics Medley
*3pBA History of High Intensity Focused Ultrasound
*3pED Acoustics Education Prize Lecture
*3pID Hot Topics in Acoustics
3pNS Sonic Boom and Numerical Methods
*3pUW Shallow Water Reverberation I

Thursday morning

*4aAAa Room Acoustics Effects on Speech Comprehension and Recall I
*4aAAb Uses, Measurements, and Advancements in the Use of Diffusion and Scattering Devices
*4aAB Use of Passive Acoustics for Estimation of Animal Population Density I
*4aBA Mechanical Tissue Fractionation by Ultrasound: Methods, Tissue Effects, and Clinical Applications I
4aEA Acoustic Transduction: Theory and Practice I
*4aPAa Borehole Acoustic Logging and Micro-Seismics for Hydrocarbon Reservoir Characterization
4aPAb Topics in Physical Acoustics I
*4aPP Physiological and Psychological Aspects of Central Auditory Processing Dysfunction I
*4aSCa Subglottal Resonances in Speech Production and Perception
4aSCb Learning and Acquisition of Speech (Poster Session)
4aSPa Imaging and Classification
4aSPb Beamforming, Spectral Estimation, and Sonar Design
*4aUW Shallow Water Reverberation II

Thursday afternoon

*4pAAa Acoustic Trick-or-Treat: Eerie Noises, Spooky Speech, and Creative Masking
*4pAAb Room Acoustics Effects on Speech Comprehension and Recall II
*4pAB Use of Passive Acoustics for Estimation of Animal Population Density II
*4pBA Mechanical Tissue Fractionation by Ultrasound: Methods, Tissue Effects, and Clinical Applications II
4pEA Acoustic Transduction: Theory and Practice II
*4pMU Assessing the Quality of Musical Instruments
*4pNS Virtual Acoustic Simulation
4pPA Topics in Physical Acoustics II
*4pPP Physiological and Psychological Aspects of Central Auditory Processing Dysfunction II
4pSC Voice (Poster Session)
*4pUW Shallow Water Reverberation III

Friday morning

*5aBA Cavitation Control and Detection Techniques
*5aED Hands-On Acoustics: Demonstrations for Indianapolis Area Students
5aNS Transportation Noise, Soundscapes, and Related Topics
5aPPa Psychological and Physiological Acoustics Potpourri (Poster Session)
5aPPb Perceptual and Physiological Mechanisms, Modeling, and Assessment
5aSC Speech Perception and Production in Challenging Conditions (Poster Session)
*5aUW Acoustics, Ocean Dynamics, and Geology of Canyons


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

	M am	M pm	M eve	Tu am	Tu pm	Tu ev	W am	W pm	W ev	Th am	Th pm	Th ev	Fri am
Indiana A/B		1pBA 1:15		2aBA 7:55	2pBA 1:30		3aBA 8:00	3pBA 1:00	TCBA 7:30	4aBA 7:55	4pBA 1:30		5aBA 8:00
Indiana C/D	1aPA 8:15	1pPA 1:15		2aPA 8:30	2pPA 1:00 2pEDa 2:45 2pEDb 3:30	TCPA 8:00	3aPA 8:20	3pED 2:00		4aPAa 8:00 4aPAb 10:30	4pPA 1:30		
Indiana E				2aNSb 8:15	2pID 1:55		3aAO 8:00	3pID 1:00					5aED 10:00
Indiana F	1aUW 8:45	1pUW 1:25		2aUW 8:00	2pUW 1:00		3aUW 9:00			4aUW 8:00	4pUW 1:00	TCUW 7:30	5aUW 8:00
Indiana G	1aSP 8:40			2aAO 8:25	2pAO 1:45	TCAO 8:00	3aSPa 8:30 8aSPb 10:15			4aSPa 9:00 4aSPb 10:15	4pAAa 1:10	TCSP 7:30	
Lincoln	1aAB 8:25	1pAB 1:00		2aAB 8:25	2pAB 1:25		3aAB 8:25			4aAB 8:00	4pAB 1:15	TCAB 7:30	
Marriott 1/2		1pSCa 1:00		2aSAa 8:00 2aSAb 10:30	2pSA 2:00	TCSA 8:00	3aSAa 8:00 3aSAb 10:00			4aPP 8:30	4pPP 1:30	TCPP 7:30	5aPPb 10:15
Marriott 3/4	1aNS 7:55	1pNS 12:55		2aNSa 9:25	2pNSa 1:25	TCSC 8:00	3aNS 8:45	3pNS 1:00			4pNS 1:15	TCNS 7:30	
Marriott 5	1aSC 9:30	1pSCb 1:00		2aSC 8:00	2pSC 1:00		3aSC 8:00			4aSCb 8:00	4pSC 1:00		5aPPa 8:00 5aSC 8:00
Marriott 6				2aED 9:00			3aID 9:00						
Marriott 7/8		1pAA 1:00		2aAA 7:55	2pAA 1:00	TCAA 8:00	3aAA 8:20	3pAA 1:00		4aAAa 8:40	4pAAb 1:15		5aNS 9:45
Marriott 9/10				2aID 8:00	2pNSb 1:00		3aEA 8:00	3pUW 1:00		4aEA 8:30	4pEA 1:30		
Santa Fe				2aMU 9:00	2pMU 1:00	TCEA 4:30	3aMU 9:00			4aSCa 8:00 4aAAb 10:35	4pMU 1:00	TCMU 7:30	
Hilbert Theater			1eID 7:00										

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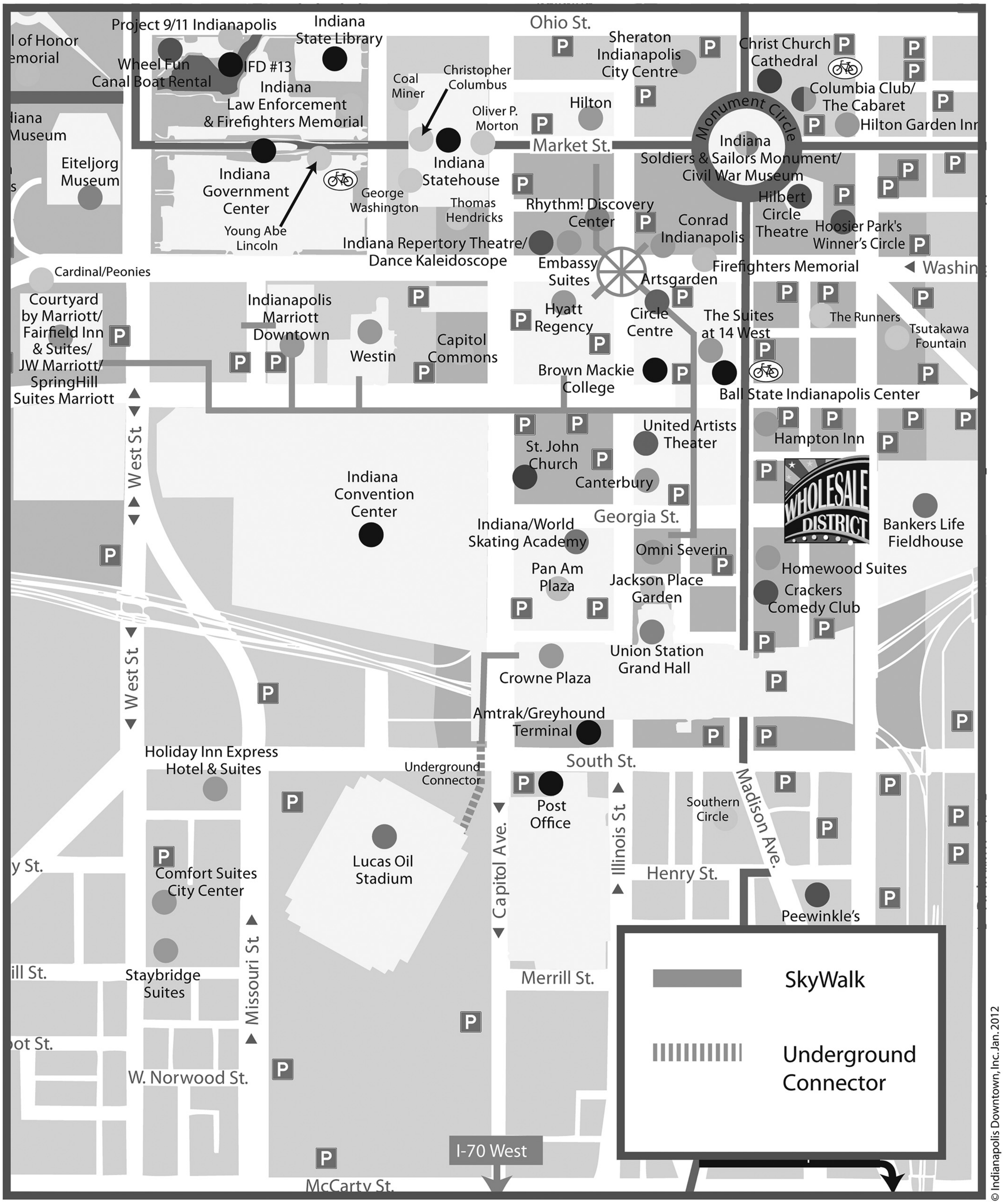


TECHNICAL PROGRAM CALENDAR
168th Meeting
Indianapolis, Indiana
27–31 October 2014

MONDAY MORNING

8:25 1aAB **Animal Bioacoustics**: Topics in Animal Bioacoustics I. Lincoln

7:55 1aNS **Noise, Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics**: Metamaterials for Noise Control I. Marriott 3/4

8:15 1aPA **Physical Acoustics and Noise**: Jet Noise Measurements and Analyses I. Indiana C/D

9:30 1aSC **Speech Communication**: Speech Processing and Technology (Poster Session). Marriott 5

8:40 1aSP **Signal Processing in Acoustics**: Sampling Methods for Bayesian Signal Processing. Indiana G

8:45 1aUW **Underwater Acoustics**: Understanding the Target/Waveguide System-Measurement and Modeling I. Indiana F

MONDAY AFTERNOON

1:00 1pAA **Architectural Acoustics**: Computer Auralization as an Aid to Acoustically Proper Owner/Architect Design Decisions. Marriott 7/8

1:00 1pAB **Animal Bioacoustics and Signal Processing in Acoustics** : Array Localization of Vocalizing Animals. Lincoln

1:15 1pBA **Biomedical Acoustics**: Medical Ultrasound. Indiana A/B

12:55 1pNS **Noise and Physical Acoustics**: Metamaterials for Noise Control II. Marriott 3/4

1:15 1pPA **Physical Acoustics and Noise**: Jet Noise Measurements and Analyses II. Indiana C/D

1:00 1pSCa **Speech Communication and Biomedical Acoustics**: Findings and Methods in Ultrasound Speech Articulation Tracking. Marriott 1/2

1:00 1pSCb **Speech Communication**: Issues in Cross Language and Dialect Perception (Poster Session). Marriott 5

1:25 1pUW **Underwater Acoustics**: Understanding the Target/Waveguide System-Measurement and Modeling II. Indiana F

MONDAY EVENING

7:00 1eID **Interdisciplinary**: Tutorial Lecture on Musical Acoustics: Science and Performance. Hilbert Theater

TUESDAY MORNING

7:55 2aAA **Architectural Acoustics and Engineering Acoustics** : Architectural Acoustics and Audio I. Marriott 7/8

8:25 2aAB **Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics**: Mobile Autonomous Platforms for Bioacoustic Sensing. Lincoln

8:25 2aAO **Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics**: Parameter Estimation in Environments that Include Out-of-Plane Propagation Effects. Indiana G

7:55 2aBA **Biomedical Acoustics**: Quantitative Ultrasound I. Indiana A/B

9:00 2aED **Education in Acoustics**: Undergraduate Research Exposition (Poster Session). Marriott 6

8:00 2aID **Archives and History and Engineering Acoustics**: Historical Transducers. Marriott 9/10

9:00 2aMU **Musical Acoustics**: Piano Acoustics. Santa Fe

9:25 2aNSa **Noise and Psychological and Physiological Acoustics**: New Frontiers in Hearing Protection I. Marriott 3/4

8:15 2aNSb **Noise and Structural Acoustics and Vibration**: Launch Vehicle Acoustics I. Indiana E

8:30 2aPA **Physical Acoustics**: Outdoor Sound Propagation. Indiana C/D

8:00 2aSAa **Structural Acoustics and Vibration and Noise**: Computational Methods in Structural Acoustics and Vibration. Marriott 1/2

10:30 2aSAb **Structural Acoustics and Vibration and Noise**: Vehicle Interior Noise. Marriott 1/2

8:00 2aSC **Speech Communication**: Speech Production and Articulation (Poster Session). Marriott 5

8:00 2aUW **Underwater Acoustics**: Signal Processing and Ambient Noise. Indiana F

TUESDAY AFTERNOON

1:00 2pAA **Architectural Acoustics and Engineering Acoustics**: Architectural Acoustics and Audio II. Marriott 7/8

1:25 2pAB **Animal Bioacoustics**: Topics in Animal Bioacoustics II. Lincoln

1:45 2pAO **Acoustical Oceanography**: General Topics in Acoustical Oceanography. Indiana G

1:30 2pBA **Biomedical Acoustics**: Quantitative Ultrasound II. Indiana A/B

2:45 2pEDa **Education in Acoustics**: General Topics in Education in Acoustics. Indiana C/D

3:30 2pEDb **Education in Acoustics**: Take 5's. Indiana C/D

1:55 2pID **Interdisciplinary**: Centennial Tribute to Leo Beranek's Contributions in Acoustics. Indiana E

1:00 2pMU **Musical Acoustics**: Synchronization Models in Musical Acoustics and Psychology. Santa Fe

1:25 2pNSa **Noise and Psychological and Physiological Acoustics**: New Frontiers in Hearing Protection II. Marriott 3/4

1:00 2pNSb **Noise and Structural Acoustics and Vibration**: Launch Vehicle Acoustics II. Marriott 9/10

1:00 2pPA **Physical Acoustics and Education in Acoustics**: Demonstrations in Acoustics. Indiana C/D

2:00 2pSA **Structural Acoustics and Vibration, Signal Processing in Acoustics, and Engineering Acoustics**: Nearfield Acoustical Holography. Marriott 1/2

1:00 2pSC **Speech Communication**: Segments and Suprasegmentals (Poster Session). Marriott 5

1:00 2pUW **Underwater Acoustics**: Propagation and Scattering. Indiana F

WEDNESDAY MORNING

8:20 3aAA **Architectural Acoustics and Noise**: Design and Performance of Office Workspaces in High Performance Buildings. Marriott 7/8

8:25 3aAB **Animal Bioacoustics**: Predator-Prey Relationships. Lincoln

8:00 3aAO **Acoustical Oceanography, Underwater Acoustics, and Education in Acoustics**: Education in Acoustical Oceanography and Underwater Acoustics. Indiana E

8:00 3aBA **Biomedical Acoustics**: Kidney Stone Lithotripsy. Indiana A/B

8:00 3aEA **Engineering Acoustics and Structural Acoustics and Vibration**: Mechanics of Continuous Media. Marriott 9/10

9:00 3aID **Student Council, Education in Acoustics, and Acoustical Oceanography**: Graduate Studies in Acoustics (Poster Session). Marriott 6

9:00 3aMU **Musical Acoustics**: Topics in Musical Acoustics. Santa Fe

8:45 3aNS **Noise and ASA Committee on Standard**: Wind Turbine Noise. Marriott 3/4

8:20 3aPA **Physical Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Noise**: Acoustics of Pile Driving: Models, Measurements, and Mitigation. Indiana C/D

8:00 3aSAa **Structural Acoustics and Vibration, Architectural Acoustics, and Noise**: Vibration Reduction in Air-Handling Systems. Marriott 1/2

10:00 3aSAb **Structural Acoustics and Vibration**: General Topics in Structural Acoustics and Vibration. Marriott 1/2

8:00 3aSC **Speech Communication**: Vowels = Space + Time, and Beyond: A Session in Honor of Diane Kewley-Port. Marriott 5

8:30 3aSPa **Signal Processing in Acoustics**: Beamforming and Source Tracking. Indiana G

10:15 3aSPb **Signal Processing in Acoustics**: Spectral Analysis, Source Tracking, and System Identification (Poster Session). Indiana G

9:00 3aUW **Underwater Acoustics, Acoustical Oceanography, Animal Bioacoustics, and ASA Committee on Standards**: Standardization of Measurement, Modeling, and Terminology of Underwater Sound. Indiana F

WEDNESDAY AFTERNOON

1:00 3pAA **Architectural Acoustics**: Architectural Acoustics Medley. Marriott 7/8

1:00 3pBA **Biomedical Acoustics**: History of High Intensity Focused Ultrasound. Indiana A/B

2:00 3pED **Education in Acoustics:** Acoustics Education Prize Lecture. Indiana C/D

1:00 3pID **Interdisciplinary**: Hot Topics in Acoustics. Indiana E

1:00 3pNS **Noise**: Sonic Boom and Numerical Methods. Marriott 3/4

1:00 3pUW **Underwater Acoustics**: Shallow Water Reverberation I. Marriott 9/10

THURSDAY MORNING

8:40 4aAAa **Architectural Acoustics, Speech Communication, and Noise**: Room Acoustics Effects on Speech Comprehension and Recall I. Marriott 7/8

10:35	4aAAb	**Architectural Acoustics**: Uses, Measurements, and Advancements in the Use of Diffusion and Scattering Devices. Santa Fe
8:00	4aAB	**Animal Bioacoustics and Acoustical Oceanography**: Use of Passive Acoustics for Estimation of Animal Population Density I. Lincoln
7:55	4aBA	**Biomedical Acoustics**: Mechanical Tissue Fractionation by Ultrasound: Methods, Tissue Effects, and Clinical Applications I. Indiana A/B
8:30	4aEA	**Engineering Acoustics**: Acoustic Transduction: Theory and Practice I. Marriott 9/10
8:00	4aPAa	**Physical Acoustics, Underwater Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, and Noise**: Borehole Acoustic Logging and Micro-Seismics for Hydrocarbon Reservoir Characterization. Indiana C/D
10:30	4aPAb	**Physical Acoustics**: Topics in Physical Acoustics I. Indiana C/D
8:30	4aPP	**Psychological and Physiological Acoustics**: Physiological and Psychological Aspects of Central Auditory Processing Dysfunction I. Marriott 1/2
8:00	4aSCa	**Speech Communication**: Subglottal Resonances in Speech Production and Perception. Santa Fe
8:00	4aSCb	**Speech Communication**: Learning and Acquisition of Speech (Poster Session). Marriott 5
9:00	4aSPa	**Signal Processing in Acoustics**: Imaging and Classification. Indiana G
10:15	4aSPb	**Signal Processing in Acoustics**: Beamforming, Spectral Estimation, and Sonar Design. Indiana G
8:00	4aUW	**Underwater Acoustics**: Shallow Water Reverberation II. Indiana F

THURSDAY AFTERNOON

1:10	4pAAa	**Architectural Acoustics and Speech Communication**: Acoustic Trick-or-Treat: Eerie Noises, Spooky Speech, and Creative Masking. Indiana G
1:15	4pAAb	**Architectural Acoustics, Speech Communication, and Noise**: Room Acoustics Effects on Speech Comprehension and Recall II. Marriott 7/8
1:15	4pAB	**Animal Bioacoustics and Acoustical Oceanography**: Use of Passive Acoustics for Estimation of Animal Population Density II. Lincoln
1:30	4pBA	**Biomedical Acoustics**: Mechanical Tissue Fractionation by Ultrasound: Methods, Tissue Effects, and Clinical Applications II. Indiana A/B
1:30	4pEA	**Engineering Acoustics**: Acoustic Transduction: Theory and Practice II. Marriott 9/10
1:00	4pMU	**Musical Acoustics**: Assessing the Quality of Musical Instruments. Santa Fe
1:15	4pNS	**Noise**: Virtual Acoustic Simulation. Marriott 3/4
1:30	4pPA	**Physical Acoustics**: Topics in Physical Acoustics II. Indiana C/D
1:30	4pPP	**Psychological and Physiological Acoustics**: Physiological and Psychological Aspects of Central Auditory Processing Dysfunction II. Marriott 1/2
1:00	4pSC	**Speech Communication**: Voice (Poster Session). Marriott 5
1:00	4pUW	**Underwater Acoustics**: Shallow Water Reverberation III. Indiana F

FRIDAY MORNING

8:00	5aBA	**Biomedical Acoustics**: Cavitation Control and Detection Techniques. Indiana A/B
10:00	5aED	**Education in Acoustics:** Hands-On Acoustics: Demonstrations for Indianapolis Area Students. Indiana E
9:45	5aNS	**Noise**: Transportation Noise, Soundscapes, and Related Topics. Marriott 7/8
8:00	5aPPa	**Psychological and Physiological Acoustics**: Psychological and Physiological Acoustics Potpourri (Poster Session). Marriott 5
10:15	5aPPb	**Psychological and Physiological Acoustics**: Perceptual and Physiological Mechanisms, Modeling, and Assessment. Marriott 1/2
8:00	5aSC	**Speech Communication**: Speech Perception and Production in Challenging Conditions (Poster Session). Marriott 5
8:00	5aUW	**Underwater Acoustics**: Acoustics, Ocean Dynamics, and Geology of Canyons. Indiana F

SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS

COUNCIL AND ADMINISTRATIVE COMMITTEES AND OTHER GROUPS

Mon, 27 Oct,7:30 a.m.	Executive Council	Denver
Mon, 27 Oct, 3:30 p.m.	Technical Council	Denver
Tue, 28 Oct, 7:00 a.m.	ASA Press Editorial Board	Illinois
Tue, 28 Oct, 7:00 a.m.	POMA Editorial Board	Denver
Tue, 28 Oct, 7:30 a.m.	Panel on Public Policy	Michigan
Tue, 28 Oct, 7:30 a.m.	Translation of Chinese Journals	Indy Boardroom
Tue, 28 Oct, 11:45 a.m.	Editorial Board	Circle City Bar & Grille
Tue, 28 Oct, 12:00 noon	Activity Kit	Illinois
Tue, 28 Oct, 12:00 noon	Prizes & Special Fellowships	Utah
Tue, 28 Oct, 12:00 noon	Student Council	Atlanta
Tue, 28 Oct, 1:30 p.m.	Meetings	Denver
Tue, 28 Oct, 4:00 p.m.	Books+	Illinois
Tue, 28 Oct, 4:00 p.m.	Education in Acoustics	Indiana C/D
Tue, 28 Oct, 4:30 p.m.	Newman Fund Advisory	Utah
Tue, 28 Oct, 5:00 p.m.	Women in Acoustics	Denver
Wed, 29 Oct, 6:45 a.m.	International Research & Education	Michigan
Wed, 29 Oct, 7:00 a.m.	College of Fellows	Florida
Wed, 29 Oct, 7:00 a.m.	Publication Policy	Illinois
Wed, 29 Oct, 7:00 a.m.	Regional Chapters	Denver
Wed, 29 Oct, 11:00 a.m.	Medals and Awards	Denver
Wed, 29 Oct, 11:15 a.m.	Public Relations	Michigan
Wed, 29 Oct, 12:00 noon	Membership	Florida
Wed, 29 Oct, 1:30 p.m.	AS Foundation Board	Illinois
Wed, 29 Oct, 5:30 p.m.	Health Care Acoustics	Utah
Thu, 30 Oct, 7:00 a.m.	Archives & History	Denver
Thu, 30 Oct, 7:00 a.m.	Tutorials	Florida
Thu, 30 Oct, 7:30 a.m.	Investment	Utah
Thu, 30 Oct, 11:00 a.m.	Acoustics Today Advisory	Illinois
Thu, 30 Oct, 2:00 p.m.	Publishing Services	Florida
Thu, 30 Oct, 4:30 p.m.	External Affairs	Michigan
Thu, 30 Oct, 4:30 p.m.	Internal Affairs	Illinois
Fri, 31 Oct, 7:00 a.m.	Technical Council	Denver
Fri, 31 Oct, 11:00 a.m.	Executive Council	Denver

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 28 Oct, 4:30 p.m.	Engineering Acoustics	Santa Fe
Tue, 28 Oct, 8:00 p.m.	Acoustical Oceanography	Indiana G
Tue, 28 Oct, 8:00 p.m.	Architectural Acoustics	Marriott 7/8
Tue, 28 Oct, 8:00 p.m.	Physical Acoustics	Indiana C/D
Tue, 28 Oct, 8:00 p.m.	Speech Communication	Marriott 3/4
Tue, 28 Oct, 8:00 p.m.	Structural Acoustics and Vibration	Marriott 1/2
Wed, 29 Oct, 7:30 p.m.	Biomedical Acoustics	Indiana A/B
Thu, 30 Oct, 7:30 p.m.	Animal Bioacoustics	Lincoln
Thu, 30 Oct, 7:30 p.m.	Musical Acoustics	Santa Fe
Thu, 30 Oct, 7:30 p.m.	Noise	Marriott 3/4
Thu, 30 Oct, 7:30 p.m.	Psychological and Physiological Acoustics	Marriott 1/2
Thu, 30 Oct, 7:30 p.m.	Signal Processing in Acoustics	Indiana G
Thu, 30 Oct, 7:30 p.m.	Underwater Acoustics	Indiana F

STANDARDS COMMITTEEES AND WORKING GROUPS

Mon, 27 Oct, 1:00 p.m.	S12/WG11-Hearing Protectors	Atlanta
Mon, 27 Oct, 7:00 p.m.	ASACOS Steering	Atlanta
Tue, 28 Oct, 7:00 a.m.	S1/WG4-Sound Pressure Levels	Atlanta
Tue, 28 Oct, 7:00 a.m.	ASACOS	Boston/Austin
Tue, 28 Oct, 4:00 p.m.	S1/WG20-Ground Impedance	Atlanta

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon-Thu, 27-30 Oct, 7:30 a.m. - 5:00 p.m.	Registration	Marriott Foyer
Fri, 31 Oct, 7:30 a.m. - 12:00 noon		
Mon-Thu, 27-30 Oct, 7:00 a.m. - 5:00 p.m.	E-mail/Internet Café	Marriott 6
Fri, 31 Oct, 7:00 a.m. - 12:00 noon		
Mon-Thu, 27-30 Oct, 7:00 a.m. - 5:00 p.m.	A/V Preview	Albany
Fri, 31 Oct, 7:00 a.m. - 12:00 noon		
Mon-Thu, 27-30 Oct, 8:00 a.m. to 10:00 a.m.	Accompanying Persons	Texas
Mon-Fri, 27-31 Oct, 9:40 a.m. - 10:40 a.m.	Coffee Break	Marriott 6
Tue-Thu, 28-30 Oct,		Indiana Ballroom Foyer
Sun, 26 Oct, 1:00 p.m. - 5:00 p.m.	Short Course	Santa Fe
Mon, 27 Oct, 7:30 a.m. - 12:30 p.m.		
Mon-Thu, 27-30 Oct, 9:00 a.m.-5:00 p.m.	Gallery of Acoustics	Marriott 6
Tue-Thu, 28-30 Oct, 12:00 noon - 1:00 p.m.	Resume Help Desk	Marriott Foyer
Mon, 27 Oct, 5:00 p.m. - 5:30 p.m.	Student Orientation	Marriott 9/10
Mon, 27 Oct, 5:30 p.m. - 6:45 p.m.	Student Meet and Greet	Marriott 6
Mon, 27 Oct, 6:00 p.m. - 7:00 p.m.	Pre-Tutorial Tour of Hilbert Circle Theater	Hilbert Circle Theater
Mon, 27 Oct, 7:00 p.m.-9:00 p.m.	Tutorial Lecture	Hilbert Circle Theater
Tue, 28 Oct, 10:00 a.m. - 12:00 noon	Tour: Center for the Performing Arts	Missouri Street Entrance
Tue, 28 Oct, 6:00 p.m. - 9:00 p.m.	Social at Eiteljorg Museum	Eiteljorg Museum
Wed, 29 Oct, 11:30 a.m. - 1:30 p.m.	Women in Acoustics Luncheon	Circle City Bar and Grille
Wed, 29 Oct, 3:30 p.m.	Annual Membership Meeting	Marriott 5
Wed, 29 Oct, 3:30 p.m. - 4:30 p.m.	Plenary Session and Awards Ceremony	Marriott 5
Wed, 29 Oct, 6:45 p.m.-8:15 p.m.	Student Reception	Indiana E
Wed, 29 Oct, 8:00 p.m. - 12:00 midnight	ASA Jam	Marriott 6
Thu, 30 Oct, 12:00 noon - 2:00 p.m.	Society Luncheon and Lecture	Indiana E
Thu, 30 Oct, 3:00 p.m. - 6:00 p.m.	Tour: 3M Acoustics Facilities	Missouri Street Entrance
Thu, 30 Oct, 6:00 p.m. - 7:30 p.m.	Social	Marriott 5/6

168th Meeting of the Acoustical Society of America

The 168th meeting of the Acoustical Society of America will be held Monday through Friday, 27–31 October 2014 at the Marriott Indianapolis Downtown Hotel, Indianapolis, Indiana, USA.

SECTION HEADINGS

1. HOTEL INFORMATION

The Indianapolis Marriott Downtown Hotel is the headquarters hotel where all meeting events will be held. Note that there are three Marriott hotels in Indianapolis so please specify the Downtown as your destination when traveling.

The cut-off date for reserving rooms at special rates has passed. Please contact the Indianapolis Marriott Downtown Hotel for reservation information: 350 West Maryland Street, Indianapolis, IN 46225, Tel: 317-822-3500.

2. TRANSPORTATION AND TRAVEL DIRECTIONS

Indianapolis is served by many major airlines through Indianapolis International Airport (IND). Information is available at www.indianapolisairport.com. The airport terminal consists of one centralized check-in area with gates on two concourses A and B which are connected via walkways to each other as well as to the check-in and reception areas of the airport. You can easily walk from the terminal via an elevated walkway to the car rental desks, which are in the Ground Transportation Center. Also, located in the same area are the limos and ground transportation information desks.

TAXI. Taxis depart from just outside the baggage claim area on the ground floor of the terminal. There is a minimum charge of USD $15 for all taxis, whatever the distance travelled. Typical costs to downtown Indianapolis are USD $15 to USD $20. Driving time to reach the Downtown Marriott is about 30 minutes.

GO GREEN LINE AIRPORT SHUTTLE. The shuttle leaves the airport on the hour and the half hour from Zone #7 on the road just outside the Ground Transportation Center. The cost is USD $10 (debit/credit card only accepted by drivers) and takes about 36 minutes to reach the Marriott complex which includes the downtown Marriott (as well as Springhill Suites, the JW Marriott, Courtyard Marriott and Fairfield Inn). You can book online at goexpresstravel.com.

BUS SERVICE TO AND FROM THE AIRPORT. IndyGo's Route 8 (www.indygo.net/maps-schedules/airport-service) provides non-express, fixed-route service from the airport to downtown via stops along Washington Street. Cost is USD $1.75 per ride. For further information and route maps visit http://www.indygo.net/maps-schedules/airport-service. The buses stop close to all the major downtown hotels. The stop at West and Washington is just northwest of the hotel. Pick-up stops are slightly different but still nearby. The stop at the airport is at Zone #6 on the road just outside the Ground Transportation Center.

SHARED-RIDE AND PERSONAL LUXURY LIMOUSINE SERVICES. These transportation services are available. Information desks are located in the Ground Transportation Center. A list of limousine companies can be found at www.indianapolisairport.com.

RENTAL CAR. Renting a car is not recommended unless you are planning trips out of town. Most everything you need should be within walking distance of the hotel. There are a lot of very nice restaurants, museums and shops reasonably close to the hotel. If you do need a rental car, the desks are located in the Ground Transportation Center on the 1st floor (ground level) of the parking garage. Alamo, Avis, Budget, Dollar, Enterprise, Hertz, National, and Thrifty all have desks at the airport and ACE has an off-airport location with a shuttle service to and from the airport, pick up just outside the Ground Transportation Center.

Amtrak and Greyhound both serve Indianapolis and the train and bus stations are within walking distance of the conference hotel. However, trains to do not run very often, e.g., one a day from Chicago to Indianapolis, versus seven a day Greyhound buses from Chicago to Indianapolis. The Amtrak station is at 350 S. Illinois Street a 10-minute walk (0.5 miles) from the Marriott and the Greyhound Station is next to the Amtrak station at 154 W. South St. See www.greyhound.com and tickets.amtrak.com for more information.

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

Messages for attendees may be left by calling the Indianapolis Marriott Downtown Hotel, 317-822-3500, and asking for the ASA Registration Desk during the meeting, where a message board will be located. This board may also be used by attendees who wish to contact one another.

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, 27 October, at 7:30 a.m. in the Marriott Ballroom Foyer on the second floor (see floor plan on page A11).

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 $545 for members of the Acoustical Society of America; $645 for non-members, $150 for Emeritus members (Emeritus status pre-approved by ASA), $275 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, $130 for students who are not members of ASA, $115 for Undergraduate Students, and $150 for accompanying persons.

One-day registration is available at $275 for members and $325 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 2015 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 $110 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@aip.org

7. TECHNICAL SESSIONS

The technical program includes 92 sessions with 948 papers scheduled for presentation during the meeting.

A floor plan of the Marriott Hotel appears on page A11. 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, 27 October
2-Tuesday, 28 October
3-Wednesday, 29 October
4-Thursday, 30 October
5-Friday, 31 October

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" or "b" 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

Hot Topics session 3pID will be held on Wednesday, 29 October, at 1:00 p.m. in Indiana E. Papers will be presented on current topics in the fields of Education in Acoustics, Signal Processing in Acoustics, and Acoustical Oceanography.

10. ROSSING PRIZE IN ACOUSTICS EDUCATION AND ACOUSTICS EDUCATION PRIZE LECTURE

The 2014 Rossing Prize in Acoustics Education will be awarded to Colin Hansen, University of Adelaide, at the Plenary Session on Wednesday, 29 October. Colin Hansen will present the Acoustics Education Prize Lecture titled "Educating mechanical engineers in the art of noise control" on Wednesday, 29 October, at 2:00 p.m. in Session 3pED in Indiana C/D.

11. TUTORIAL LECTURE: MUSICAL ACOUSTICS: SCIENCE AND PERFORMANCE

A tutorial presentation on "Musical Acoustics: Science and Performance" will be given by Professor Uwe J. Hansen of Indiana State University, and the New World Youth Symphony, directed by Susan Kitterman, on Monday, 27 October at 7:00 p.m. in the Hilbert Circle Theater.

The Tutorial Concert will be preceded by a tour of Hilbert Circle Theater, home of the Indianapolis Symphony. Hilbert Circle Theater was a movie house. It underwent major revisions to make it suitable as a concert hall. Since the last ASA meeting in Indianapolis in 1996, the concert hall has undergone additional major remodeling, mainly in the stage area, but also in the hall itself. The tour will begin at 6:00 p.m.

Hilbert Circle Theater is well within easy walking distance of the hotel (allow 15 minutes to get there), however, in the event of inclement weather, and for those with additional needs, limited bus transportation will be available (from 5:30pm onwards).

Lecture notes will be available at the meeting in limited supply; only preregistrants will be guaranteed receipt of a set of notes.

All Students (K – Grad school) will be admitted free of charge. General admission, for both the general Public and ASA members is USD $20.00. ASA members who include attendance in this tutorial concert in their pre-registration by 22 September pay the reduced fee of USD $15.00.

12. SHORT COURSE ON ELECTROACOUSTIC TRANSDUCERS

A short course on Electroacoustic Transducers: Fundamentals and Applications will be given in two parts: Sunday, 26 October, from 1:00 p.m. to 5:00 p.m. and Monday, 27 October, from 7:30 a.m. to 12:30 p.m. in the Santa Fe Room.

The objectives are (1) to introduce the physical principles, basic performance, and system design aspects required for effective application of receiving and transmitting transducers and (2) to present common problems and potential solutions.

The instructor is Thomas Gabrielson, a Senior Scientist and Professor of Acoustics at Penn State University, previously worked in underwater-acoustic transducer design, modeling, and measurement for 22 years at the Naval Air Warfare Center in Warminster, PA.

The registration fee is USD$300.00 (USD$125 for students) and covers attendance, instructional materials and coffee breaks. Onsite registration at the meeting will be on a space-available basis.

13. UNDERGRADUATE RESEARCH POSTER EXPOSITION

The Undergraduate Research Exposition will be held Tuesday morning, 28 October, 9:00 a.m. to 11:00 a.m. in session 2aED in Marriott 6. The 2014 Undergraduate Research Exposition is a forum for undergraduate students to present their research pertaining to any area of acoustics and can also include overview papers on undergraduate research programs, designed to inspire and foster growth of undergraduate research throughout the Society. It is intended to encourage undergraduates to express their knowledge and interest in acoustics and foster their participation in the Society. Four awards, up to $500 each, will be made to help undergraduates with travel costs associated with attending the meeting and presenting a poster.

14. 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 Marriott Ballroom Foyer near the registration desk. Members of the ASA experienced in hiring will be available to look at your CV, 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 for walk-up meetings. Appointments during these three lunch hours will be available via a sign-up sheet, too.

15. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday at the Indianapolis Marriott Downtown. The meetings on Tuesday and Thursday will be held in the evenings after the socials, except Engineering Acoustics which will meet at 4:30 p.m. on Tuesday. The schedule and rooms for each Committee meeting are given on page A16.

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.

16. TECHNICAL TOURS

Note: Tour buses leave from the Marriott's Missouri Street Exit.

Monday, 27 October, 6:00 p.m.-8:30 p.m. Tour and Tutorial Lecture at the Hilbert Circle Theater, 45 Monument Circle, Indianapolis. Tour fees: USD $15 preregistration and USD $20 on-site for non-students/No fee for students. Prior to becoming the home of the Indianapolis Symphony, Hilbert Circle Theater was a movie house. It underwent major revisions to make it suitable as a concert hall. Tour starts at 6:00 p.m. at the theater and tutorial presentation starts at 7:00 p.m. The Theater is half a mile walking distance (about 15 minutes from the hotel, so leave at 5:45 p.m. at the latest). See the Tutorial Lecture section above for full details.

Tuesday, 28 October: 10:00 a.m.-12:00 noon. Tour of the Center for the Performing Arts, 355 City Center Drive, Carmel. Tour limited to 30 participants. Tour fee: USD $25. The Center for Performing Arts houses the Palladium (1,600 seat concert hall), the Tarkington Theater (500 seats proscenium stage) and the Studio Theater (small flexible black box space). This is a recently completed facility north of Indianapolis. The Palladium is a space that rivals the world's great concert halls. David M. Schwarz Architects, a Washington, DC based architectural firm, drew inspiration for the Palladium from the famous Villa Capra "La Rotunda" (or Villa Rotunda) built in 1566 in Italy and designed by Italian Renaissance architect Andrea Palladio (1508–1580). For more information about the Center visit www.thecenterfortheperformingarts.org.

Tuesday, 28 October: 3:00 p.m.-6:00 p.m. Indiana University School of Medicine, 699 Riley Hospital Drive, Indianapolis. Tour limited to 30 participants. Tour fee: USD $25. The Department of Otolaryngology-Head and Neck Surgery was organized as an independent department within the Indiana School of Medicine in 1909 by John Barnhill, M.D., an internationally recognized head and neck surgeon and anatomist. Since then the specialty has undergone tremendous expansion in managing disorders of the ear, nose, throat, head and neck. The DeVault Otologic Research Laboratory is the primary behavioral research venue for the Department. Occupying approximately 3000 square feet on two floors of the research wing of the James Whitcomb Riley Hospital for Children, the laboratory is named for its principal early benefactor, Dr. Virgil T. DeVault (1901-2000), a native Hoosier and alumnus of Indiana University. In the laboratory researchers examine the short-term and long-term effects of cochlear implantation and/or therapeutic amplification in deaf and hard-of-hearing infants, children, and adults, as well as the factors underlying variability in behavioral outcomes of cochlear implantation and/or therapeutic amplification.

Thursday, 30 October: 3:00 p.m.-6:00 p.m. Tour of 3M Acoustics Facilities, 7911 Zionsville Road, Indianapolis. Tour limited to 30 participants. Tour fee: USD $25 Elliott Berger and Steve Sorenson will give tours of 3M's E•A•RCAL hearing protection laboratory and Acoustic Technology Center (ATC) laboratory for noise control research and application. The E•A•RCAL facility consists of a NVLAP accredited 113-m^3 reverberation chamber instrumented for real-ear attenuation testing, an 18-m^3 electroacoustic sound lab supporting high-level tests up to 120 dB SPL, and a 300-m^3 hemi-anechoic facility used for impulse testing via a shock tube that generates blasts up to 168 dB SPL for measuring the level-dependent performance of hearing protectors. The ATC includes a 900-m^3 hemi-anechoic chamber with an inbuilt chassis dynamometer ideal for testing heavy trucks under real-world load conditions, a smaller hemi-anechoic chamber for product sound power testing, and 2 reverberation chambers for a wide variety of sound transmission loss and sound absorption test/development. Note that no photographs are allowed to be taken on this tour. Conference attendees who work for 3M/Aearo/E-A-R competitors may not be allowed to participate in the tour – the registration fee will be refunded in full should the request to participate (through pre-registration) not be approved.

Start times are when the bus leaves the hotel, so plan on being there ahead of time.

On-site registration will be on a space-available basis.

17. GALLERY OF ACOUSTICS

The Technical Committee on Signal Processing in Acoustics will sponsor the 15th Gallery of Acoustics at the Acoustical Society of America meeting in Indianapolis. 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 of Acoustics provides a means by which we can all share and appreciate the natural beauty, aesthetic, and artistic appeal of acoustic phenomena: This is a forum where science meets art.

The Gallery will be held in the Marriott Ballroom 6, Monday through Thursday, 27-30 October, from 9:00 a.m. to 5:00 p.m.

18. ANNUAL MEMBERSHIP MEETING

The Annual Membership Meeting of the Acoustical Society of America will be held at 3:30 p.m. on Wednesday, 29 October 2014, in Marriott 5 at the at the Indianapolis Downtown Marriott Hotel, 350 West Maryland Street, Indianapolis, IN 46225.

19. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 29 October, at 3:30 p.m. in Marriott 5.

The Rossing Prize in Acoustics Education will be presented to Colin Hansen. The Pioneers of Underwater Acoustics Medal will be presented to Michael B. Porter, the Silver Medal in Speech Communication will be presented to Sheila E. Blumstein and the Wallace Clement Sabine Medal will be presented to Ning Xiang. Certificates will be presented to Fellows elected at the Providence meeting of the Society. See page 2228 for a list of fellows.

20. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will not be held at the Indianapolis meeting.

Meetings of selected advisory working groups are often held in conjunction with Society meetings and are listed in the Schedule of Committee Meetings and Other Events on page A16 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@aip.org

21. COFFEE BREAKS

Morning coffee breaks will be held each day from 9:40 a.m. to 10:40 a.m. in Marriott 6.

22. A/V PREVIEW ROOM

The Albany Room on the second floor 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 Indianapolis meeting will have a published proceedings, and submission is optional. The proceedings will be a separate volume of the online journal, "Proceedings of Meetings on Acoustics" (POMA). 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. The format requirements for POMA are somewhat more stringent than for posting on the ASA Online Meetings Papers Site, but the two versions could be the same. The posting at the Online Meetings Papers site, however, is not archival, and posted papers will be taken down six months after the meeting. The POMA online site for submission of papers from the meeting will be opened about one-month after authors are notified that their papers have been accepted for presentation. It is not necessary to wait until after the meeting to submit one's paper to POMA. Further information regarding POMA can be found at the site http://asadl/poma/for_authors_poma. Published papers from previous meeting can be seen at the site http://asadl/poma.

24. E-MAIL ACCESS, INTERNET CAFÉ, AND BREAK ROOM

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 Marriott 6.

A unique feature of this ASA meeting is that a ballroom located directly opposite the registration area will be dedicated as a central gathering area for discussion, wi-fi, coffee breaks, the Gallery of Acoustics and more. Join your colleagues the Break Room every day to discuss the latest ASA topics and news.

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

25. SOCIALS

The Eiteljorg Museum of American Indian and Western art will be the site for the social on Tuesday, October 28, from 6:00 p.m. to 9:00 p.m. Galleries will be open for viewing art that promotes understanding of the history and cultures of North American people, including their contemporary Native art collection that has been ranked among the world's best. The collections are housed in a striking building, located along the White River Canal within easy walking distance (about 6 minutes) of the Indianapolis Marriott Downtown. For those who prefer not to walk, shuttle service will be available throughout the evening to and from the Missouri Street exit of the hotel. In keeping with the Museum, the reception will feature a delectable array of food selections having a slightly Southwestern flair.

A Halloween Social for all, even noisy spirits or eerie creatures, will take place on Thursday, October 30 in the Marriott Ballroom from 6:00 p.m. to 7:30 p.m. Costumes are positively encouraged, so don't forget to pack one. Get ready for a few fun surprises organized by a team of young acousticians that are sure to provide some great photo ops. To set the stage for Thursday night's activities, Halloween fun is included in a Thursday afternoon technical session sponsored by Architectural Acoustics and Speech Communication. In this session other worldly minds offer 13 talks from 1:00 p.m. to 5:00 p.m. in Marriott Ballroom 5/6. Come to learn about the acoustics of supernatural spirits, bumps in the night, eerie voices and other sorts of spooky audition.

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, 30 October, at 12:00 noon in Indiana E. The luncheon is open to all attendees and their guests. The speaker is Larry E. Humes, Distinguished Professor and Department Chair, Department of Speech and Hearing Sciences, Indiana University. Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 29 October. The cost is $30.00 per ticket.

27. STUDENTS MEET MEMBERS FOR LUNCH

The ASA Education Committee arranges for a student to meet one-on-one with a member of the Acoustical Society over lunch. The purpose is to make it easier for students to meet and interact with members at ASA Meetings. Each lunch pairing is arranged separately. Students who are interested should contact Dr. David Blackstock, University of Texas at Austin, by email dtb@mail.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, 27 October, in Marriott 9/10 for all students to learn about the activities and opportunities available for students at the Indianapolis ASA meeting. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in Marriott 6. Refreshments and a cash bar will be available. Students are encouraged to attend the tutorial lecture on which begins at 7:00 p.m. in The Hilbert Theater. Student registration for this event is free.

The Students' Reception will be held on Wednesday, 29 October, from 6:45 p.m. to 8:15 p.m. in Indiana E. 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 www.acosoc.org/student/ to learn more about student involvement in ASA.

29. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 29 October, in the Circle City Bar and Grille on the first floor of the Marriott. Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 28 October. The fee is USD$30 for non-students and USD$15 for students.

30. JAM SESSION

You are invited to Marriott 6 on Wednesday night, 29 October, in Marriott 6 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. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Indianapolis meeting. The on-site registration fee for accompanying persons is USD$150. A hospitality room for accompanying persons will be open in the Texas Room at the Indianapolis Marriott Downtown Hotel from 8:00 a.m. to 10:00 a.m. Monday through Thursday. For updates about the accompanying persons program please check the ASA website at AcousticalSociety.org/meetings.html.

Visit: **http://visitindy.com** to learn about what is going on in Indianapolis. Good places to visit within walking distance include the Eiteljorg Museum (Tuesday night social venue), the Indiana State Museum (with IMAX theater), the NCAA Hall of Champions, the Indianapolis Zoo, and White River State Park (you can hire bikes there). Further away requiring transportation (taxi or bus - **http://www.indygo.net/pages/system-map**) is the Indianapolis Speedway Museum which is at the Indy 500 track, the Children's Museum, and the Indiana Museum of Art. Close to the hotel, there is the Circle Center Mall which is a great place for shopping.

32. WEATHER

Weather in Indianapolis in the last week in October can vary a lot from year to year. Make sure you are prepared for rain so you can take full advantage of nearby restaurants and attractions. See http://visitindy.com and the hotel website http://www.marriott.com/hotels/travel-guide/indcc-indianapolis-marriott for more information about Indianapolis. There is a 35% chance of some sort of precipitation (rain) and snow is very rare at that time of year. Average low and high temperatures at that time of year are 41 and 60 degrees F, respectively.

33. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Robert F. Port, Chair; David R. Dowling, Acoustical Oceanography; Roderick J. Suthers, Animal Bioacoustics; Norman H. Philipp, Architectural Acoustics; Robert J. McGough, Biomedical Acoustics; Uwe J. Hansen, Education in Acoustics; Roger T. Richards, Engineering Acoustics; Andrew C.H. Morrison, Musical Acoustics; William J. Murphy, Noise; Kai Ming Li, Physical Acoustics; Jennifer Lentz, Psychological and Physiological Acoustics; R. Lee Culver, Cameron Fackler, Signal Processing in Acoustics; Diane Kewley-Port, Alexander L. Francis, Speech Communication; Benjamin M. Shafer, Structural Acoustics and Vibration; Kevin L Williams, Underwater Acoustics.

34. MEETING ORGANIZING COMMITTEE

Kenneth de Jong and Patricia Davies, Cochairs; Robert F. Port, Technical Program Chair; Diane Kewley-Port, Tessa Bent, Mary C. Morgan, Food and Beverage; Mary C. Morgan, Kai Ming Li, Tom Lorenzen, Audio-Visual and WiFi; Caroline Richie, Volunteer Coordination; William J. Murphy, Technical Tours; Uwe Hansen, Educational Activities, Tutorials; Diane Kewley-Port, Special Events; Maria Kondaurova, Guanguan Li, Michael Hayward, Indianapolis Visitor Information; Tessa Bent, Student Activities; Mary C. Morgan, Meeting Administrator.

35. PHOTOGRAPHING AND RECORDING

Photographing and recording during regular sessions are not permitted without prior permission from the Acoustical Society.

36. ABSTRACT ERRATA

This meeting program is Part 2 of the October 2014 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*.

37. 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 slide (e.g., Powerpoint). 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 institutions 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.

38. 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 Acknowledgments

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 sheets) to distribute to interested audience members.

39. 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.

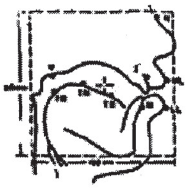
Linux

- 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.

40. 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@aip.org

169th Meeting, Pittsburgh, Pennsylvania, 18–22 May 2015
170th Meeting, Jacksonville, Florida, 2–6 November 2015
171st Meeting, Salt Lake City, Utah, 23–27 May 2016
172nd Meeting, Honolulu, Hawaii, 28 November–2 December 2016.



Session 1aAB

Animal Bioacoustics: Topics in Animal Bioacoustics I

James A. Simmons, Chair
Neuroscience, Brown University, 185 Meeting St., Box GL-N, Providence, RI 02912

Chair's Introduction—8:25

Contributed Papers

8:30

1aAB1. Spinner dolphins (*Stenella longirostris* GRAY, 1828) acoustic parameters recorded in the Western South Atlantic Ocean. Juliana R. Moron, Artur Andriolo (Instituto de Ciências Biológicas, Universidade Federal de Juiz de Fora, Rua Batista de Oliveira 1110 apto 404 B, Juiz de Fora 36010520, Brazil, julianamoron@hotmail.com), and Marcos Rossi-Santos (Centro de Ciências Agrárias, Ambientais e Biológicas, Universidade Federal do Recôncavo da Bahia, Cruz das Almas, Brazil)

Spinner dolphins bioacoustics were study only in Fernando de Noronha Archipelago region in the Western South Atlantic Ocean. Our study aimed to describe the acoustic parameters of this species recorded approximately 3500 km south of Fernando de Noronha Archipelago. An one-element hydrophone was towed 250 m behind the vessel R/V Atlântico Sul over the continental shelf break. Continuous mono recording was performed with the hydrophone passing signals to a digital Fostex® FR-2 LE, recording at 96 kHz/24 bits. A group of approximate 400 dolphins were recorded on June 3, 2013, at 168.9 km shore distance (27o 24'29"S, 46o50'05"W). The wav-files were analyzed through the spectrogram configured as DFT 512 samples, 50% overlap and Hamming window of 1024 points generated by software Raven Pro 1.4. The preliminary results of 10 min recording allowed the extraction of 693 whistles that were classified in contours shapes as: upsweep (42%), chirp (17.3%), downsweep (14%), sinusoidal (10.5%), convex (5.9%), constant (5.4%), and concave (4.9%). Minimum frequencies ranged from 3.32 kHz to 23.30 kHz (mean = 10.88 kHz); maximum frequencies ranged from 6.61 kHz to 35.34 kHz (mean = 15.77 kHz); whistle duration ranged from 0.03 s to 2.58 s (mean = 0.68 s). These results are important to understand populations and/or species distributed in different ocean basins.

8:45

1aAB2. A new method for detection of North Atlantic right whale up-calls. Mahdi Esfahanian, Hanqi Zhuang, and Nurgun Erdol (Comput. and Elec. Eng. and Comput. Sci., Florida Atlantic Univ., 777 Glades Rd., Bldg: EE 96, Rm. 409, Boca Raton, FL 33431, mesfahan@fau.edu)

A study of detecting North Atlantic Right Whale (NARW) up-calls has been conducted with measurements from passive acoustic monitoring devices. Denoising and normalization algorithms are applied to remove local variance and narrowband noise in order to isolate the NARW up-calls in spectrograms. The resulting spectrograms, after binarization, are treated with a region detection procedure called the Moor-Neighbor algorithm to find continuous objects that are candidates of up-call contours. After selected properties of each detected object are computed, they are compared with a pair of low and high empirical thresholds to estimate the probability of the detected object being an up-call; therefore, those objects that are determined with certainty to be non up-calls are discarded. The final stage in the proposed call detection method is to separate true up-calls from the rest of potential up-calls with classifiers such as linear discriminate analysis (LDA), Naïve Bayes, and decision tree. Experimental results using the data set obtained by Cornell University show that the proposed method can achieve accuracy to 96%.

9:00

1aAB3. Spatio-temporal distribution of beaked whales in southern California waters. Simone Baumann-Pickering, Jennifer S. Trickey (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), and Sean M. Wiggins (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Cuvier's beaked whales are the dominant beaked whales offshore of southern California. Their abundance, distribution, and seasonality are poorly understood. Insights on the spatio-temporal distribution of both Cuvier's and a rare beaked whale with signal type BW43, likely Perrin's beaked whale, have been derived from long-term autonomous recordings of beaked whale echolocation clicks. Acoustic recordings were collected at 18 sites offshore of southern California since 2006, resulting in a total of ~26 years of recordings. About 23,000 acoustic encounters with Cuvier's beaked whales were detected. In contrast, there were ~100 acoustic encounters of the BW43 signal type. Cuvier's beaked whales were predominantly detected at deeper, more southern, and farther offshore sites, and there appears to be a seasonal pattern to their presence, with lower probability of detection during summer and early fall. The BW43 signal type had higher detection rates in the central basins, indicating a possible difference in habitat preference and niche separation between the two species. Further investigation is needed to reveal if these distribution patterns are purely based on bathymetric preference, driven by water masses that determine prey species composition and distribution, or possibly by anthropogenic activity.

9:15

1aAB4. The acoustic characteristics of greater prairie-chicken vocalizations. Cara Whalen, Mary Bomberger Brown (School of Natural Resources, Univ. of Nebraska - Lincoln, 3310 Holdrege St., Lincoln, NE 68583, cara-whalen@gmail.com), JoAnn McGee (Developmental Auditory Physiol. Lab., Boys Town National Res. Hospital, Omaha, NE), Larkin A. Powell, Jennifer A. Smith (School of Natural Resources, Univ. of Nebraska - Lincoln, Lincoln, NE), and Edward J. Walsh (Developmental Auditory Physiol. Lab., Boys Town National Res. Hospital, Omaha, NE)

Male Greater Prairie-Chickens (*Tympanuchus cupido pinnatus*) congregate in groups known as "leks" each spring to perform vocal and visual displays to attract females. Four widely recognized vocalization types produced by males occupying leks are referred to as "booms," "cackles," "whines," and "whoops." As part of a larger effort to determine the influence of wind turbine farm noise on lek vocal behavior, we studied the acoustic properties of vocalizations recorded between March and June in 2013 and 2014 at leks near Ainsworth, Nebraska. Although all four calls are produced by males occupying leks, the boom is generally regarded as the dominant call type associated with courtship behavior. Our findings suggest that the bulk of acoustic power carried by boom vocalizations is in a relatively narrow, low frequency band, approximately 100-Hz wide at 20 dB below the peak frequency centered on approximately 0.3 kHz. The boom vocalization is harmonic in character, has a fundamental frequency of

approximately 0.30±0.01 kHz, and lasts approximately 1.81±0.18 s. Understanding Greater Prairie-Chicken vocal attributes is an essential element in the effort to understand the influence of environmental sound, prominently including anthropogenic sources like wind turbine farms, on vocal communication success.

9:30

1aAB5. Bioacoustics of *Trachymyrmex fuscus*, *Trachymyrmex tucumanus,* and *Atta sexdens rubropilosa* (Hymenoptera: Formicidae). Amanda A. Carlos, Francesca Barbero, Luca P. Cassaci, Simona Bonelli (Life Sci. and System Biology, Univ. of Turin, Dipartimento di Biologia Animale e dell'Uomo Via Accademia Albertina 13, Turin 10123, Italy, amandacarlos@yahoo.com.br), and Odair C. Bueno (Centro de Estudos de Insetos Sociais (CEIS), Universidade Estadual Paulista Júlio de Mesquita Filho (UNESP), Rio Claro, Brazil)

The capability to produce species-specific sounds is common among ants. Ants of the genus *Trachymyrmex* occur in an intermediate phylogenetic position within the Attini tribe, between the leafcutters, such as *Atta sexdens rubropilosa*, and more basal species. The study of stridulations would provide important cues on the evolution of the tribe's diverse biological aspects. Therefore, in the present study, we described the stridulation signals produced by *Trachymyrmex fuscus*, *Trachymyrmex tucumanus,* and *A. sexdens rubropilosa* workers. Ant workers were recorded, and their stridulatory organs were measured. The following parameters were analyzed: chirp length [ms], inter-chirp (pause) [ms], cycle (chirp + inter-chirp) [ms], cycle repetition rate [Hz], and the peak frequency [Hz], as well as the number of ridges on the *pars stridens*. During the inter-chirp, there is no measurable signal for *A. sexdens rubropilosa*, whereas for *Trachymyrmex fuscus* and *Trachymyrmex tucumanus*, a low intensity signal was detected. In other words, the plectrum and the *pars stridens* of *A. sexdens rubropilosa* have no contact during the lowering of the gaster. Principal component analysis, to which mainly the duration of chirps contributed, showed that stridulation is an efficient tool to differentiate ant species at least in the case of the Attini tribe.

9:45

1aAB6. Robustness of perceptual features used for passive acoustic classification of cetaceans to the ocean environment. 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 cetaceans in their habitats, which cover diverse underwater environments. It is well known that properties of the ocean environment can be markedly different between regions, which can result in distinct propagation characteristics. These can in turn lead to differences in the time-frequency characteristics of a recorded signal and may impact the accuracy of PAM systems. To develop an automatic PAM system capable of operating under numerous environmental conditions, one must account for the impact of propagation conditions. A prototype aural classifier developed at Defence R&D Canada 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 uses a combination of at-sea experiments and pulse propagation modeling to examine the robustness of the perceptual features with respect to propagation effects. Preliminary results will be presented from bowhead and humpback vocalizations that were transmitted over 1–20 km ranges during a two-day sea trial in the Gulf of Mexico. Insight gained from experimental results will be augmented with model results. [Work supported by the U.S. Office of Naval Research.]

10:00–10:15 Break

10:15

1aAB7. Passive acoustic monitoring on the seasonal species composition of cetaceans from a marine observatory. Tzu-Hao Lin, Hsin-Yi Yu (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, schonkopf@gmail.com), Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), and Lien-Siang Chou (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan)

Information on the species diversity of cetaceans can help us to understand the community ecology of marine top predators. Passive acoustic monitoring has been widely applied in the cetacean research, however, species identification based on tonal sounds remains challenging. In order to examine the seasonal changing pattern of species diversity, we applied an automatic detection and classification algorithm on acoustic recordings collected from the marine cable hosted observatory off the northeastern Taiwan. Representative frequencies of cetacean tonal sounds were detected. Statistical features were extracted based on the distribution of representative frequency and were used to classify four cetacean groups. The correct classification rate was 72.2% based on the field recordings collected from onboard surveys. Analysis on one-year recordings revealed that the species diversity was highest in winter and spring. Short finned pilot whales and Risso's dolphins were the most common species, they mainly occurred in winter and summer. False killer whales were mostly detected in winter and spring. Spinner dolphins, spotted dolphins, and Fraser's dolphins were mainly detected in summer. Bottlenose dolphins represent the least common species. In the future, the biodiversity, species-specific habitat use, and inter-specific interaction of cetaceans can be investigated through an underwater acoustic monitoring network.

10:30

1aAB8. The effects of road noise on the calling behavior of Pacific chorus frogs. Danielle V. Nelson (Dept. of Forest Ecosystems and Society, Oregon State Univ., Oregon State University, 321 Richardson Hall, Corvallis, OR 97331, danielle.nelson@oregonstate.edu), Holger Klinck (Fisheries and Wildlife, Oregon State Univ., Newport, OR), and Tiffany S. Garcia (Fisheries and Wildlife, Oregon State Univ., Corvallis, OR)

Fitness consequences of anthropogenic noise on organisms that have chorus-dependent breeding requirements, such as frogs, are not well understood. While frogs were thought to have innate and fixed call structure, species-specific vocal plasticity has been observed in populations experiencing high noise conditions. Adjustment to call structure, however, can have negative fitness implications in terms of energy expenditure and female choice. The Pacific chorus frog (*Pseudacris regilla*), a common vocal species broadly distributed throughout the Pacific Northwest, often breeds in waters impacted by road noise. We compared Pacific chorus frog call structure from breeding populations at 11 high- and low-traffic sites in the Willamette Valley, Oregon. We used passive acoustic monitoring and directional recordings to determine mean dominant frequency, amplitude, and call rate of breeding populations, individual frogs, and to quantify ambient road noise levels. Preliminary results indicate that while individuals do not differ in call rate or structure across noisy and quiet sites, high road noise levels decrease the effective communication distance of both the chorus and the individual. This research enhances our understanding of acoustic habitat in the Willamette Valley and the impacts of anthropogenic noise on a native amphibian species.

10:45

1aAB9. Inter-individual difference of one type of pulsed sounds produced by beluga whales (*Delphinapterus leucas*). Yuka Mishima (Tokyo Univ. of Marine Sci. and Technol., Konan 4-5-7, Minato-ku, Tokyo 108-8477, Japan, thank_you_for_email_5yuka@yahoo.co.jp), Tadamichi Morisaka (Tokai Univ. Inst. of Innovative Sci. and Technol., Shizuoka-shi, Japan), Miho Itoh (The Port of Nagoya Public Aquarium, Nagoya-shi, Japan), Ryota Suzuki, Kenji Okutsu (Yokohama Hakkeijima Sea Paradise, Yokohama-shi, Japan), Aiko Sakaguchi, and Yoshinori Miyamoto (Tokyo Univ. of Marine Sci. and Technol., Minato-ku, Japan)

Belugas often exchange one type of broadband pulsed sounds (termed PS1 calls) which possibly functions as a contact calls (Morisaka *et al.*,

2013). Here we investigate how belugas embed their signature information into the PS1 calls. PS1 calls were recorded from each of five belugas including both sexes and various ages at the Port of Nagoya Public Aquarium using a broadband recording system when in isolation. Temporal and spectral acoustic parameters of PS1 calls were measured and compared among individuals. Kruskal-Wallis test revealed that inter-pulse intervals (IPIs), the number of pulses, and pulse rates of PS1 calls had significant differences among individuals, but duration did not ($\chi 2 = 76.7$, $p < 0.0001$; $\chi 2 = 26.2$, $p < 0.0001$; $\chi 2 = 45.3$, $p < 0.0001$; and $\chi 2 = 4.7$, $p = 0.316$ respectively). The contours depicted by the IPIs as a function of pulse order were also individually different and only the contours of a calf fluctuated over time. Four belugas except a juvenile had individually distinctive power spectra. These results suggest that several acoustic parameters of PS1 calls may hold individual information. We found PS1-like calls from the other captive belugas (Yokohama Hakkeijima Sea Paradise) suggested that the PS1 call is not the specific call for one captive population but the basic call type for belugas.

11:00

1aAB10. Numerical study of biosonar beam forming in finless porpoise (*Neophocaena asiaeorientalis*). Chong Wei (College of Ocean & Earth Sci., Xiamen Univ., 1502 Spreckels St. Apt 302A, Honolulu, Hawaii 96822, weichong3310@foxmail.com), Zhitao Wang (Key Lab. of Aquatic Biodiversity and Conservation of the Chinese Acad. of Sci., Inst. of Hydrobiology of the Chinese Acad. of Sci., Wuhan, China), Zhongchang Song (College of Ocean & Earth Sci., Xiamen Univ., Xiamen, China), Whitlow Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, Kaneohe, HI), Ding Wang (Key Lab. of Aquatic Biodiversity and Conservation of the Chinese Acad. of Sci., Inst. of Hydrobiology of the Chinese Acad. of Sci., Wuhan, China), and Yu Zhang (Key Lab. of Underwater Acoust. Commun. and Marine Information Technol. of the Ministry of Education, Xiamen Univ., Xiamen, China)

Finless porpoise (*Neophocaena asiaeorientalis*) is known to use the narrow band signals for echolocation living in the Yangtze River and in the adjoining Poyang and Dongting Lakes in China. In this study, the sound velocity and density of different tissues (including melon, muscle, bony structure, connective tissues, blubber, and mandibular fat) in the porpoise's head were obtained by measurement. The sound velocity and density were found out to have a linear relationship with Hounsfield unit (HU) obtained from the CT scan. The acoustic property of the head of the porpoise was reconstructed from the HU distribution. Numerical simulations of the acoustic propagation through finless porpoise's head were performed by a finite element approach. The beam formation was compared with those of the baiji, Indo-pacific humpback dolphin, and bottlenose dolphin. The role of the different structures in the head such as air sacs, melon, muscle, bony structure, connective tissues, blubber, and mandibular fat on biosonar beam was investigated. The results might provide useful information for better understanding of the sound propagation in finless porpoise's head.

11:15

1aAB11. Evidence for a possible functional significance of horseshoe bat biosonar dynamics. Rolf Müller, Anupam K. Gupta (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, rolf.mueller@vt.edu), Uzair Gillani (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Yanqing Fu (Eng. Sci. and Mech., Virginia Tech, Blacksburg, VA), and Hongxiao Zhu (Dept. of Statistics, Virginia Tech, Blacksburg, VA)

The periphery of the biosonar system of horseshoe bats is characterized by a conspicuous dynamics where the shapes of the noseleaves (structures that surround the nostrils) and the outer ears (pinnae) undergo fast changes that can coincide with pulse emission and echo reception. These changes in the geometries of the sound-reflecting surfaces affect the device characteristics, e.g., as represented by beampatterns. Hence, this dynamics could give horseshoe bats an opportunity to view their environments through a set of different device characteristics. It is not clear at present whether horseshoe bats make use of this opportunity, but there is evidence from various sources, namely, anatomy, behavior, evolution, and information theory. Anatomical studies have shown the existence of specialized muscular actuation systems that are clearly directed toward geometrical changes. Behavioral observations indicate that these changes are linked to contexts where the bat is confronted with a novel or otherwise demanding situation. Evolutionary evidence comes from the occurrence of qualitatively similar ear deformation patterns in mustached bats (*Pteronotus*) that have independently evolved a biosonar for Doppler-shift detection. Finally, an information-theoretic analysis demonstrates that the capacity of the biosonar system for encoding sensory information is enhanced by these dynamic processes.

11:30

1aAB12. Analysis of some special buzz clicks. Odile Gerard (DGA, Ave. de la Tour Royale, Toulon 83000, France, odigea@gmail.com), Craig Carthel, and Stefano Coraluppi (Systems & Technol. Res., Woburn, MA)

Toothed whales are known to click regularly to find prey. Once a prey has been detected, the repetition rate of the clicks increases; these sequences are called buzzes. Previous work shows that the buzz clicks spectrum slowly varies from click to click for various species. This spectrum similarity allows buzz clicks association as a sequence using multi-hypothesis tracking (MHT) algorithms. Thus buzz classification follows automatic click tracking. The use of MHT reveals that in some rare cases a variant of this property has been found, whereby sub-sequences of clicks exhibit slowly varying characteristics. In 2010 and 2011, NATO Undersea Research Centre (NURC, now CMRE Centre for Maritime Research and Experimentation) conducted sea-trials with the CPAM (compact Passive Acoustic Monitoring), a volumetric towed array comprised of four or six hydrophones. This configuration allows for a rough estimate of clicking animal localization. Some buzzes with sub-sequences of slowly varying characteristics were recorded with the CPAM. Localization may help to understand this new finding from a physiological point of view. The results of this analysis will be presented.

Session 1aNS

Noise, Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Metamaterials for Noise Control I

Keith Attenborough, Cochair
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Olga Umnova, Cochair
University of Salford, The Crescent, Salford M5 4WT, United Kingdom

Chair's Introduction—7:55

Invited Papers

8:00

1aNS1. Recent results on sonic crystals for sound guiding and acoustic absorption. Jose Sanchez-Dehesa, Victor M. García-Chocano, and Matthew D. Guild (Dept. of Electron. Eng., Universitat Politecnica de Valencia, Camino de vera s.n., Edificio 7F, Valencia, Valencia ES-46022, Spain, jsdehesa@upv.es)

We report on different aspects of the behavior of sonic crystals with finite size. First, at wavelengths on the order of the lattice period we have observed the excitation of deaf modes, i.e., modes with symmetry orthogonal to that of the exciting beam. Numerical simulations and experiments performed with samples made of three rows of cylindrical scatterers demonstrate the excitation of sound waves guided along a direction perpendicular to the incident beam. Moreover, the wave propagation inside the sonic crystal is strongly dependent on the porosity of the building units. This finding can be used to enhance the absorbing properties of the crystal. Also, we will discuss the properties of finite sonic crystals at low frequencies, where we have observed small period oscillations superimposed on the well-known Fabry-Perot resonances appearing in the reflectance and transmittance spectra. It will be shown that the additional oscillations are due to diffraction in combination with the excitation of the transverse modes associated with the finite size of the samples. [Work supported by ONR.]

8:20

1aNS2. Acoustic metamaterial absorbers based on multi-scale sonic crystals. Matthew D. Guild, Victor M. García-Chocano (Dept. of Electronics Eng., Universitat Politecnica de Valencia, Camino de vera s/n (Edificio 7F), Valencia 46022, Spain, mdguild@utexas.edu), Weiwei Kan (Dept. of Phys., Nanjing Univ., Nanjing, China), and Jose Sanchez-Dehesa (Dept. of Electronics Eng., Universitat Politecnica de Valencia, Valencia, Spain)

In this work, thermoviscous losses in single- and multi-scale sonic crystal arrangements are examined, enabling the fabrication and characterization of acoustic metamaterial absorbers. It will be shown that higher filling fraction arrangements can be used to provide a large enhancement in the complex mass density and loss factor, and can be combined with other sonic crystals of different sizes to create multi-scale structures that further enhance these effects. To realize these enhanced properties, different sonic crystal lattices are examined and arranged as a layered structure or a slab with large embedded inclusions. The inclusions are made from either a single solid cylinder or symmetrically arranged clusters of cylinders, known as magic clusters, which behave as an effective fluid. Theoretical results are obtained using a two-step homogenization process, by first homogenizing each sonic crystal to obtain the complex effective properties of each length scale, and then homogenizing the effective fluid structures to determine the properties of the ensemble structure. Experimental data from acoustic impedance tube measurements will be presented and shown to be in excellent agreement with the expected results. [Work supported by the US ONR and Spanish MINECO.]

8:40

1aNS3. Quasi-flat acoustic absorber enhanced by metamaterials. Abdelhalim Azbaid El Ouahabi, Victor V. Krylov, and Daniel J. O'Boy (Dept. of Aeronautical and Automotive Eng., Loughborough Univ., Loughborough University, Loughborough, Leicestershire LE11 3TU, United Kingdom, A.Azbaid-El-Ouahabi@lboro.ac.uk)

In this paper, the design of a new quasi-flat acoustic absorber (QFAA) enhanced by the presence of a graded metamaterial layer is described, and the results of the experimental investigation into the reflection of sound from such an absorber are reported. The matching metamaterial layer is formed by a quasi-periodic array of brass cylindrical tubes with the diameters gradually increasing from the external row of tubes facing the open air towards the internal row facing the absorbing layer made of a porous material. The QFAA is placed in a wooden box with the dimensions of 569 × 250 × 305 mm. All brass tubes are of the same length (305 mm) and fixed between the opposite sides of the wooden box. Measurements of the sound reflection coefficients from the empty wooden box, from the box with an inserted porous absorbing layer, and from the full QFAA containing both the porous absorbing layer and the array of brass tubes have

been carried out in an anechoic chamber at the frequency range of 500–3000 Hz. The results show that the presence of the metamaterial layer brings a noticeable reduction in the sound reflection coefficients in comparison with the reflection from the porous layer alone.

9:00

1aNS4. The influence of thermal and viscous effects on the effective properties of an array of slits. John D. Smith (Physical Sci., DSTL, Porton Down, Salisbury SP4 0JQ, United Kingdom, jdsmith@dstl.gov.uk), Roy Sambles, Gareth P. Ward, and Alastair R. Murray (Dept. of Phys. and Astronomy, Univ. of Exeter, Exeter, United Kingdom)

A system consisting of an array of thin plates separated by air gaps is examined using the method of asymptotic homogenization. The effective properties are compared with a finite element model and experimental results for the resonant transmission of both a single slit and an array of slits. These results show a dramatic reduction in the frequency of resonant transmission when the slit is narrowed to below around one percent of the wavelength due to viscous and thermal effects reducing the effective sound velocity through the slits. These effects are still significant for slit widths substantially greater than the thickness of the boundary layer.

9:20

1aNS5. Atypical dynamic behavior of periodic frame structures with local resonance. Stéphane Hans, Claude Boutin (LGCB / LTDS, ENTPE / Université de Lyon, rue Maurice Audin, Vaulx-en-Velin 69120, France, stephane.hans@entpe.fr), and Céline Chesnais (IFSTTAR GER, Université Paris-Est, Paris, France)

This work investigates the dynamic behavior of periodic unbraced frame structures made up of interconnected beams or plates. Such structures can represent an idealization of numerous reticulated systems, as the microstructure of foams, plants, bones, the sandwich panels. As beams are much stiffer in tension-compression than in bending, the propagation of waves with wavelengths much greater than the cell size and the bending modes of the elements can occur in the same frequency range. Thus, frame structures can behave as metamaterials. Since the condition of scale separation is respected, the homogenization method of periodic discrete media is used to derive the macroscopic behavior. The main advantages of the method are the analytical formulation and the possibility to study the behavior of the elements at local scale. This provides a clear understanding of the mechanisms governing the dynamics of the material. In the presence of the local resonance, the form of the equations is unchanged but some macroscopic parameters depend on the frequency. In particular, this applies to the mass leading to a generalization of the Newtonian mechanics. As a result, there are frequency bandgaps. In that case, the same macroscopic modal shape is also associated with several resonant frequencies.

9:40

1aNS6. Design of sound absorbing metamaterials by periodically embedding three-dimensional resonant or non-resonant inclusions in rigidly backed porous plate. Jean-Philippe Groby (LAUM, UMR6613 CNRS, LAUM, UMR 6613 CNRS, AV. Olivier Messiaen, Le Mans F-72085, France, Jean-Philippe.Groby@univ-lemans.fr), Benoit Nennig (LISMMA, Supmeca, Saint Ouen, France), Clément Lagarrigue, Brunuo Brouard, Dazel Olivier (LAUM, UMR6613 CNRS, Le Mans, France), Olga Umnova (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), and Vincent Tournat (LAUM, UMR6613 CNRS, Le Mans, France)

Air saturated porous materials, namely, foams and wools, are often used as sound absorbing materials. Nevertheless, they suffer from a lack of absorption efficiency at low frequencies, which is inherent to their absorption mechanisms (viscous and thermal losses), even when used as optimized multilayer or graded porous materials. These last decades, several solutions have been proposed to avoid this problem. Among them, metaporous materials consist in exciting modes trapping the energy between the periodic rigid inclusions embedded in the porous plate and the rigid backing or in the inclusions themselves. The absorption coefficient of different foams is enhanced both in the viscous and inertial regimes by periodically embedding 3D inclusions, possibly resonant, i.e., air filled Helmholtz resonators. This enhancement is due to different mode excitation: a Helmholtz resonance in the viscous regime and a trap mode in the inertial regime. In particular, a large absorption coefficient is reached for wavelengths in the air 27 times larger than the sample thickness. The absorption amplitude and bandwidth is then enlarged by removing porous material in front of the neck, enabling a lower impedance radiation, and by adjusting the resonance frequencies of the Helmholtz resonator.

10:00–10:20 Break

10:20

1aNS7. Seismic metamaterials: Shielding and focusing surface elastic waves in structured soils. Sebastien R. Guenneau, Stefan Enoch (Phys., Institut Fresnel, Ave. Escadrille Normandie Niemen, Marseille 13013, France, sebastien.guenneau@fresnel.fr), and Stephane Brule (Menard Co., Nozay, France)

Phononic crystals and metamaterials are man-made structures (with periodic heterogeneities typically a few micrometers to centimeters) that can control sound in ways not found in nature. Whereas the properties of phononic crystals derive from the periodicity of their structure, those of metamaterials arise from the collective effect of a large array of small resonators. These effects can be used to manipulate acoustic waves in unconventional ways, realizing functions such as invisibility cloaking, subwavelength focusing, and unconventional refraction phenomena (such as negative refractive index and phase velocity). Recent work has started to explore another intriguing domain of application: using similar concepts to control the propagation of seismic waves within the surface of the Earth. Our research group at the Aix-Marseille University and French National Center for Scientific Research (CNRS) has teamed up with civil engineers at an industrial company, Ménard, in Nozay, also in France, and carried out the largest-scale tests to date of phononic crystals. Arrays of boreholes in soil which are a few centimeters to a few meters in diameter are encouraging, thereafter called seismic metamaterials, can be used to deflect incoming acoustic waves at a frequency relevant to earthquake protection, or bring them to a focus. These preliminary successes could one day translate into a way of mitigating the destructive effects of earthquakes.

10:40

1aNS8. Tunable resonator arrays—Transmission, near-field interactions, and effective property extraction. Dmitry Smirnov and Olga Umnova (Acoust. Res. Ctr., Univ. of Salford, The Crescent, Salford, Greater Manchester m5 4wt, United Kingdom, d.smirnov@edu.salford.ac.uk)

Periodic arrays of slotted cylinders have been studied with a focus on analytical and semi-analytical techniques, observing near-field interactions and their influence on reflection and transmission of acoustic waves by the array. Relative orientation of the cylinders within a unit cell has been shown to strongly affect the array behavior, facilitating tunable transmission gaps. An improved homogenization method is proposed and used to determine effective properties of the array, allowing accurate and computationally efficient prediction of reflection and transmission characteristics of any number of rows at arbitrary incidence.

11:00

1aNS9. Tunable cylinders for sound control in water. Andrew Norris and Alexey Titovich (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Long wavelength effective medium properties are achieved using arrays of closely spaced tunable cylinders. Thin metal shells provide the starting point: for a given shell thickness h and radius a, the effective bulk modulus and density are both proportional to h/a. Since the metal has large impedance relative to water it follows that there is a unique value of h/a at which the shell is effectively impedance matched to water. The effective sound speed cannot be matched by the thin shell alone (except for impractical metals like silver). However, simultaneous impedance and speed matching can be obtained by adding an internal mass, e.g., an acrylic core in aluminum cylindrical tubes. By varying the shell thickness and the internal mass, a range of effective properties is achievable. Practical considerations such as shell thickness, internal mass material, and fabrication will be discussed. Arrays made of a small number of different tuned shells will be described using numerical simulations: example applications include focusing, lensing, and wave steering. [Work supported by ONR.]

11:20

1aNS10. Sound waves over periodic and aperiodic arrays of cylinders on ground surfaces. Shahram Taherzadeh, Ho-Chul Shin, and Keith Attenborough (Eng. & Innovation, The Open Univ., Walton Hall, Milton Keynes MK7 6AA, United Kingdom, shahram.taherzadeh@open.ac.uk)

Propagation of audio frequency sound waves over periodic arrays of cylinders placed on acoustically hard and soft surfaces has been studied through laboratory measurements and predictions using a point source. It is found that perturbation of the position of the cylinders from a regular array results in a higher insertion loss than completely periodic or random cylinder arrangements.

11:40

1aNS11. Ground effect due to rough and resonant surfaces. Keith Attenborough (Eng. and Innovation, Open Univ., 18 Milebush, Linslade, Leighton Buzzard, Bedfordshire LU7 2UB, United Kingdom, Keith.Attenborough@open.ac.uk), Ho-Chul Shin, and Shahram Taherzadeh (Eng. and Innovation, Open Univ., Milton Keynes, United Kingdom)

Particularly if the ground surface between noise source and receiver would otherwise be smooth and acoustically hard, structured low-rise ground roughness can be used as an alternative to conventional noise barriers. The techniques of periodic-spacing, absorptive covering, and local resonance can be used, as when broadening metamaterial stop bands, to achieve a broadband ground effect. This has been demonstrated both numerically and through laboratory experiments. Computations have employed multiple scattering theory, the Finite Element Method and the Boundary Element Method. The experiments have involved measurements over cylindrical and rectangular roughness elements and over their resonant counterparts created by incorporating slit-like openings. Resonant elements with slit openings have been found numerically and experimentally to add a destructive interference below the first roughness-induced destructive interference and thereby mitigate the adverse effects of the low-frequency surface waves generated by the presence of roughness elements. A nested configuration of slotted hollow roughness elements is predicted to produce multiple resonances and this idea has been validated through laboratory experiments.

Session 1aPA

Physical Acoustics and Noise: Jet Noise Measurements and Analyses I

Richard L. McKinley, Cochair
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Chair's Introduction—8:15

Invited Papers

8:20

1aPA1. F-35A and F-35B aircraft ground run-up acoustic emissions. Michael M. James, Micah Downing, Alexandria R. Salton (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, michael.james@blueridgeresearch.com), Kent L. Gee, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Richard L. McKinley, Alan T. Wall, and Hilary L. Gallagher (Air Force Res. Lab., Dayton, OH)

A multi-organizational effort led by the Air Force Research Laboratory conducted acoustic emissions measurements on the F-35A and F 35B aircraft at Edwards Air Force Base, California, in September 2013. These measurements followed American National Standards Institute/Acoustical Society of America S12.75-2012 to collect noise data for community noise models and noise exposures to aircraft personnel. In total, over 200 unique locations were measured with over 300 high fidelity microphones. Multiple microphone arrays were deployed in three orientations: circular arcs, linear offsets from the jet-axis centerline, and linear offsets from the jet shear layer. The microphone arrays ranged from distances 10 ft outside the shear layer to 4000 ft from the aircraft with angular positions ranging from 0° (aircraft nose) to 160° (edge of the exhaust flow field). A description of the ground run-up acoustic measurements, data processing, and the resultant data set is provided.

8:40

1aPA2. Measurement of acoustic emissions from F-35B vertical landing operations. Micah Downing, Michael James (Blue Ridge Res. and Consulting, 29 N. Market St., Ste. 700, Asheville, NC 28801, micah.downing@blueridgeresearch.com), Kent Gee, Brent Reichman (Brigham Young Univ., Provo, UT), Richard McKinley (Air Force Res. Lab., Wright-Patterson AFB, OH), and Allan Aubert (Naval Air Warfare Ctr., Patuxent River, MD)

A multi-organizational effort led by the Air Force Research Laboratory conducted acoustic emissions measurements from vertical landing operations of the F-35B aircraft at Marine Corps Air Station Yuma, Arizona, in September 2013. These measurements followed American National Standards Institute/Acoustical Society of American S12.75-2012 to collect noise data from vertical landing operations for community noise models and noise exposures to aircraft personnel. Three circular arcs and two vertical microphone arrays were deployed for these measurements. The circular microphone arrays ranged from distances from 250 ft to 1000 ft from touch down point. A description of the vertical landing acoustic measurements, data processing, preliminary data analysis, the resultant dataset, and a summary of results will be provided.

9:00

1aPA3. Acoustic emissions from flyover measurements of F-35A and F-35B aircraft. Richard L. McKinley, Alan T. Wall, Hilary L. Gallagher (Battlespace Acoust. Branch, Air Force Res. Lab., 711 HPW/RHCB, 2610 Seventh St., Bldg 441, Wright-Patterson AFB, OH, richard.mckinley.1@us.af.mil), Christopher M. Hobbs, Juliet A. Page, and Joseph J. Czech (Wyle Labs., Inc., Arlington, VA)

Acoustic emissions of F-35A and F-35B aircraft flyovers were measured in September 2013, in a multi-organizational effort led by the Air Force Research Laboratory. These measurements followed American National Standards Institute/Acoustical Society of America S12.75-2012 guidance on aircraft flyover noise measurements. Measurements were made from locations directly under the flight path to 12,000 ft away with microphones on the ground, 5 ft, and 30 ft high. Vertical microphone arrays suspended from cranes measured noise from on the ground up to 300 ft above the ground. A linear ground-based microphone array measured noise directly along the flight path. In total, data were collected at more than 100 unique locations. Measurements were repeated six times for each flyover condition. Preliminary results are presented to demonstrate the repeatability of noise data over measurement repetitions, assess data quality, and quantify community noise exposure models.

9:20

1aPA4. Three-stream jet noise measurements and predictions. Brenda S. Henderson (Acoust., NASA, MS 54-3, 21000 Brookpark Rd., Cleveland, OH 44135, brenda.s.henderson@nasa.gov) and Stewart J. Leib (Ohio Aerosp. Inst., Cleveland, OH)

An experimental and numerical investigation of the noise produced by high-subsonic three-stream jets was conducted. The exhaust system consisted of externally mixed-convergent nozzles and an external plug. Bypass- and tertiary-to-core area ratios between 1 and 1.75, and 0.4 and 1.0, respectively, were studied. Axisymmetric and offset tertiary nozzles were investigated for heated and unheated conditions. For axisymmetric configurations, the addition of the third stream was found to reduce mid- and high-frequency acoustic levels in the peak-jet-noise direction, with greater reductions at the lower bypass-to-core area ratios. The addition of the third stream also decreased peak acoustic levels in the peak-jet-noise direction for intermediate bypass-to-core area ratios. For the offset configurations, an s-duct was found to increase acoustic levels relative to those of the equivalent axisymmetric-three-stream jet while half-duct configurations produced acoustic levels similar to those for the axisymmetric jet for azimuthal observation locations of interest. Comparisons of noise predictions with acoustic data are presented for selected unheated configurations. The predictions are based on an acoustic analogy approach with mean flow interaction effects accounted for using a Green's function, computed in terms of its coupled azimuthal modes, and a source model previously used for round and rectangular jets.

9:40

1aPA5. Acoustic interaction of turbofan exhaust with deflected control surface for blended wing body airplane. Dimitri Papamoschou (Mech. and Aerosp. Eng., Univ. of California, Irvine, 4200 Eng. Gateway, Irvine, CA 92697-3975, dpapamos@uci.edu) and Salvador Mayoral (Mech. and Aerosp. Eng., Univ. of California, Irvine, Irvine, Armed Forces Pacific)

Small-scale experiments simulated the elevon-induced jet scrubbing noise of the Blended-Wing-Body platform with a bypass ratio ten turbofan nozzle installed above the wing. The elevon chord length at the interaction zone was similar to the exit fan diameter of the nozzle. The study encompassed variable nozzle position, variable elevon deflection, removable inboard fins, and two types of nozzles—plain and chevron. Far-field microphone surveys were conducted underneath the wing. The interaction between the jet and the elevon produces excess noise that intensifies with increasing elevon deflection. When the elevon trailing edge is near the edge of the jet, excess noise is manifested as a low-frequency bump on the sound pressure level spectrum. An empirical model for this excess noise is presented. The interaction noise becomes severe, and elevates the entire spectrum, when the elevon intrudes significantly into the jet flow. The increase in effective perceived noise level (EPNL) falls on well-defined trends when correlated versus the penetration of the elevon trailing edge into the flow field of the isolated jet. The cumulative takeoff EPNL can increase by as much as 19 dB, underscoring the potentially detrimental effects of jet-elevon interaction on noise compliance.

10:00–10:20 Break

10:20

1aPA6. Comparison of upside-down microphone with flush mounted microphone configuration. Per Rasmussen (G.R.A.S. Sound & Vib. A/S, Skovlytoften 33, Holte 2840, Denmark, pr@gras.dk)

Measurement of fly-over aircraft noise is often performed using the microphones mounted in an upside-down configuration, with the microphone placed 7 mm above a hard reflecting surface. This method assumes that most of the sound is coming from the back of the microphone within an angle of + -60 degrees. The same microphone configuration is proposed for installed and un-installed jet-engine test in which case, however, the incidence angle for the microphone may be in the range of 60–85 degrees. The response of the upside-down microphone configuration is compared with flush mounted microphones as reference. The influence of microphone diameter (ranging from 1/8 in. to ½ in.) is compared in the different configurations and the effect of windscreens is investigated.

10:40

1aPA7. Active control of noise from hot, supersonic turbulent jets. Tim Colonius, Aaron Towne (Mech. Eng., Caltech, 1200 E. California Blvd., Pasadena, CA 91125, colonius@caltech.edu), Robert H. Schlinker, Ramons A. Reba, and Dan Shannon (Thermal and Fluid Sci. Dept., United Technologies Res. Ctr., East Hartford, CT)

We report on an experimental and reduced-order modeling study aimed at reducing mixing noise in hot supersonic jets relevant to military aircraft. A spinning valve is used to modulate four injection nozzles near the main jet nozzle lip over a range of frequencies and mass flow rates. Diagnostics include near-, mid-, and far-field microphone arrays aimed at measuring the effect of actuation on the near-field turbulent wavepacket structures and their correlation with mixing noise. The actuators provide more than 4 dB noise reduction at peak frequencies in the aft arc, and up to 2 dB reduction in OASPL. Experiments are performed to contrast the performance of steady and unsteady blowing with different amplitudes. The results to date suggest that the noise reduction is primarily associated with attenuated wave packet activity associated with the rapidly thickened shear layers that occur with both steady and unsteady blowing. Mean flow surveys are also preformed and serve as inputs to reduced-order models for the wave packets based on parabolized stability equations. These models are in turn used to corroborate the experimental evidence suggesting mechanisms of noise suppression in the actuated flow.

Contributed Papers

1a MON. AM

11:00

1aPA8. Efficient jet noise models using the one-way Euler equations. Aaron Towne and Tim Colonius (Dept. of Mech. and Civil Eng., California Inst. of Technol., 1200 E California Blvd., MC 107-81, Pasadena, CA 91125, atowne@caltech.edu)

Experimental and numerical investigations have correlated large-scale coherent structures in turbulent jets with acoustic radiation to downstream angles, where sound is most intense. These structures can be modeled as linear instability modes of the turbulent mean flow. The parabolized stability equations have been successfully used to estimate the near-field evolution of these modes, but are unable to properly capture the acoustic field. We have recently developed an efficient method for calculating these linear modes that properly captures the acoustic field. The linearized Euler equations are modified such that all upstream propagating acoustic modes are removed from the operator. The resulting equations, called one-way Euler equations, can be stably and efficiently solved in the frequency domain as a spatial initial value problem in which initial perturbations are specified at the flow inlet and propagated downstream by integrating the equations. We demonstrate the accuracy and efficiency of the method by using it to model sound generation and propagation in jets. The results are compared to accurate large-eddy-simulation data for both subsonic and supersonic jets.

11:15

1aPA9. A new method of estimating acoustic intensity applied to the sound field near a military jet aircraft. Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen, Derek C. Thomas, Benjamin Y. Christensen (Phys. and Astronomy, Brigham Young Univ., 688 north 500 East, Provo, UT 84606, tstout@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting LLC, Asheville, NC)

Intensity probes are traditionally made up of closely spaced microphones, with the finite-difference method used to approximate acoustic intensity. This approximation is not reliable approaching the Nyquist frequency limit determined by microphone spacing. However, the new phase and amplitude estimation (PAGE) method allows for accurate intensity approximation far above this limit. The PAGE method is applied to measurements from a three-dimensional intensity probe, which took data to the sideline and aft of a tethered F-22A Raptor. It is shown that the PAGE method produces physically meaningful intensity approximations for frequencies up to about 6 kHz, while the finite-difference method is only reliable up to about 2 kHz. [Work supported by ONR.]

11:30

1aPA10. Three transformations of a crackling jet noise waveform and their potential implications for quantifying the "crackle" percept. S. Hales Swift (School of Aeronautics and Astronautics, Purdue Univ., 2286 Yeager Rd., West Lafayette, IN 47906, hales.swift@gmail.com), Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In the 1975 paper by Ffowcs-Williams *et al.* on jet "crackle," there are several potentially competing descriptors—including a qualitative description of the sound quality or percept, a statistical measure, and commentary on the relation of the presence of shocks to the sound's quality. These descriptors have led to disparate conclusions about what constitutes a crackling jet, waveform, or sound quality. This presentation considers three modifications of a jet noise waveform that exhibits a crackling sound quality and initially satisfies all three definitions. These modifications alter the statistical distributions of primarily the pressure waveform or its first time difference in order to demonstrate how these modifications do or do not correspond to changes in the sound quality of the waveform. The result, although preliminary, demonstrates that the crackle percept is tied to the statistics of the pressure difference waveform instead of the pressure waveform itself.

MONDAY MORNING, 27 OCTOBER 2014 — MARRIOTT 5, 9:30 A.M. TO 12:00 NOON

Session 1aSC

Speech Communication: Speech Processing and Technology (Poster Session)

Michael Kiefte, Chair
Human Communication Disorders, Dalhousie University, 1256 Barrington St., Halifax, NS B3J 1Y6, Canada

All posters will be on display from 9:30 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 9:30 a.m. to 10:45 a.m. and contributors of even-numbered papers will be at their posters from 10:45 a.m. to 12:00 noon.

Contributed Papers

1aSC1. Locus equations estimated form a corpus of running speech. Michael Kiefte (Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca) and Terrance M. Nearey (Linguist, Univ. of AB, Edmonton, NS, Canada)

Locus equations, or the linear relationship between onset and vowel second-formant frequency F2 in terms of slope and y-intercept, have been presented as possible invariant correlates to consonant place of articulation [e.g., Sussman *et al.* (1998). Behav. Brain Sci. 21, 241–299]. In the current study, formant measurements were extracted from both stressed and unstressed vowels taken from a database of spontaneous and read speech. Locus equations were estimated for several places of articulation of the preceding consonant. In addition, optimal time frames for estimating locus equations are determined with reference to automatic classification of consonant place of articulation as well as vowel identification. Formant frequencies are first measured at multiple time frames—both before and after voicing onset in the case of voiceless plosives—to find the pair of time frames that best estimates place of articulation via discriminant analysis and

other classification methods. In addition, locus-equation slopes are compared between stressed and unstressed vowels as well as between spontaneous and read speech samples. In addition, the role of total vowel duration across these contexts is described. The evaluation of several strategies for optimizing the automatic extraction of formant frequencies from running speech are also reported. [Work supported by SSHRC.]

1aSC2. Formant trajectory analysis using dynamic time warping: Preliminary results. Kirsten T. Regier (Linguist, Indiana Univ., 3201 W Woodbridge Dr., Muncie, IN 47304, krtodt@indiana.edu)

In English, there are at least two mechanisms that affect vowel duration—vowel identity and postvocalic consonant voicing. Previous studies have shown that these two mechanisms have independent effects on vowel duration (Port 1981, Todt 2010). This study presents preliminary results on the use of dynamic time warping to distinguish between the effects of vowel identity and postvocalic consonant voicing on the formant trajectories of English front vowels. Using PraatR (Albin 2014), formant trajectories are extracted from sound files in Praat and imported into R, where the dynamic time warping analysis is conducted using the dtw package (Giorgino 2009). Albin, A. L. (2014). PraatR: An architecture for controlling the phonetics software "Praat" with the R programming language. JASA 135, 2198. Giorgino T. (2009). "Computing and Visualizing Dynamic Time Warping Alignments in R: The dtw Package,"J. Stat. Software, 31(7), pp. 1–24. Port, R. F. (1981). Linguistic timing factors in combination. JASA 69(1), 262–274. R Core Team (2014). R: A language and environment for statistical computing. R Foundation for Statistical Computing, Vienna, Austria. Todt, K. R. (2010). The production of English front vowels by Spanish speakers: A study of vowel duration based on vowel tenseness and consonant voicing, JASA 128, 2489.

1aSC3. A "pivot" model for extracting formant measurements based on vowel trajectory dynamics. Aaron L. Albin and Wil A. Rankinen (Dept. of Linguist, Indiana Univ., Memorial Hall 322, 1021 E 3rd St., Bloomington, IN 47405-7005, aaalbin@indiana.edu)

Formant measurements are commonly extracted at fixed fractions across a vowel's duration (e.g., the 1/2 point for a monophthong and the 1/3 and 2/3 points for a diphthong). This approach tacitly relies on the convenience assumption that a speaker always maximally approximates the intended acoustic target at roughly the same point across a vowel's duration. The present paper proposes an alternate method whereby every formant point sampled within a vowel is considered as a possible "pivot" (i.e., turning point), with monophthongs modeled as having one pivot and diphthongs modeled as having two pivots. The optimal pivot for the vowel is then determined by fitting regression lines to the formant trajectory and comparing the goodness-of-fit of these lines to the raw formant data. When applied to a corpus of an American English dialect, the resulting measurements were found to be significantly correlated with previous methods. This suggests that the aforementioned convenience assumption is unnecessary and that the proposed model, which is more faithful to our understanding of articulatory dynamics, is a viable alternative. Moreover, rather than being assumed a priori, the location of the measurement can be treated as an empirical question in its own right.

1aSC4. Exploiting second-order statistics improves statistical learning of vowels. Fernando Llanos (School of Lang. and Cultures, Purdue Univ., 220 FERRY ST APT 6, Lafayette, IN 45901, fllanos@purdue.edu), Yue Jiang, and Keith R. Kluender (Dept. of Speech, Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN)

Unsupervised clustering algorithms were used to evaluate three models of statistical learning of minimal contrasts between English vowel pairs. The first two models employed only first-order statistics with assumptions of uniform [M1] or Gaussian [M2] distributions of vowels in an F1-F2 space. The third model [M3] employed second-order statistics by encoding covariance between F1 and F2. Acoustic measures of F1/F2 frequencies for 12 vowels spoken by 139 men, women, and children (Hillendrand *et al.* 1995) were used as input to the models. Effectiveness of each model was tested for each minimal-pair contrast across 100 simulations. Each simulation consisted of two centroids that adjusted on a trial-by-trial basis as 1000 F1/F2 pairs were input to the models. With addition of each pair, centroids were reallocated by a k-means algorithm, an unsupervised clustering algorithm that provides an optimal partition of the space into uniformly-sized convex cells. The first-order Gaussian model [M2] performed better than a uniform distribution [M2] for six of seven minimal pairs. The second-order model [M3] was significantly superior to both first-order models for every pair. Results have implications for optimal perceptual learning of phonetic differences in ways that respect lawful covariance across vocal tract lengths that vary across talkers.

1aSC5. Analysis of acoustic to articulatory speech inversion for natural speech. Ganesh Sivaraman (Elec. & Comput. Eng., Univ. of Maryland College Park, 7704 Adelphi Rd., Apt 11, Hyattsville, MD 20783, ganesa90@umd.edu), Carol Espy-Wilson (Elec. & Comput. Eng., Univ. of Maryland College Park, College Park, MD), Vikramjit Mitra (SRI Int.., Menlo Park, CA), Hosung Nam (Korea Univ., Seoul, South Korea), and Elliot Saltzman (Physical Therapy & Athletic Training, Boston Univ., New Haven, Connecticut)

Speech inversion is a technique to estimate vocal tract configurations from speech acoustics. We constructed two such systems using feedforward neural networks. One was trained using natural speech data from the XRMB database and the second using synthetic data generated by the Haskins Laboratories TADA model that approximated the XRMB data. XRMB pellet trajectories were first converted into vocal tract constriction variables (TVs), providing a relative measure of constriction kinematics (location and degree) and synthetic TV data was obtained directly using TADA. The natural and synthetic speech inversion systems were trained as TV estimators using these respective sets of acoustic and TV data. TV-estimators were first tested using previously collected acoustic data on the utterance "perfect memory" spoken at slow, normal, and fast rates. The TV estimator trained on XRMB data (but not on TADA data) was able to recover the tongue tip gesture for /t/ in the fast utterance despite the gesture occurring partly during the acoustic silence of the closure. Further, the XRMB system (but not the TADA system) could distinguish between bunched and retroflexed /r/. Finally, we compared the performance of the XRMB system with a set of independently trained speaker-dependent systems (using the XRMB database) to understand the role of speaker-specific differences in the partitioning of variability across acoustic and articulatory spaces.

1aSC6. Testing AutoTrace: A machine-learning approach to automated tongue contour data extraction. Gustave V. Hahn-Powell (Linguist, Univ. of Arizona, 2850 N Alvernon Way, Apt 17, Tucson, AZ 85712, hahnpowell@email.arizona.edu) and Diana Archangeli (Linguist, Univ. of Hong Kong, Tucson, Arizona)

While ultrasound provides a remarkable tool for tracking the tongue's movements during speech, it has yet to emerge as the powerful research tool it could be. A major roadblock is that the means of appropriately labeling images is a laborious, time-intensive undertaking. In earlier work, Fasel and Berry (2010) introduced a "translational" deep belief network (tDBN) approach to automated labeling of ultrasound images of the tongue, and tested it against a single-speaker set of 3209 images. This study tests the same methodology against a much larger data set (about 40,000 images), using data collected for different studies with multiple speakers and multiple languages. Retraining a "generic" network with a small set of the most erroneously labeled images from language-specific development sets resulted in an almost three-fold increase in precision in the three test cases examined.

1aSC7. Usability of SpeechMark® landmark analysis system for teaching speech acoustics. Marisha Speights and Suzanne E. Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, PO Box 670379, Cincinnati, OH 45267-0379, speighma@mail.uc.edu)

Learning about the intersection of articulation and acoustics, and particularly acoustic measurement techniques, is challenging for students in Linguistics, Psychology and Communication Sciences and Disorders curricula. There is a steep learning curve before students can apply the material to an interesting research question; for those in more applied programs such as

Communication Disorders or ESL, there is an additional challenge in envisioning how the knowledge can be applied in changing behavior. The availability of software tools such as Wavesurfer, Praat, Audacity, TF32, the University College of London software suite, among others, has made it possible for instructors to design laboratory experiences in visualization, manipulation, and measurement of speech acoustics. Many students have found them complex for their first exposure to taking scientific measurements. The SpeechMark® acoustic landmark analysis system has been developed to automate the detection of specific acoustic events important for speech, such as voicing offset and onset, stop bursts, fricative noise, and vowel midpoints, and to provide automated formant frequency measurement used for vowel space analysis. This paper describes a qualitative multiple case study in which seven teachers of speech acoustics were interviewed to explore whether such pre-analysis of the acoustic signal could be useful for teaching.

1aSC8. Surveying the nasal peak: A1 and P0 in nasal and nasalized vowels. Will Styler and Rebecca Scarborough (Linguist, Univ. of Colorado, 295 UCB, Boulder, CO 80309, william.styler@colorado.edu)

Nasality can be measured in the acoustical signal using A1-P0, where A1 is the amplitude of the harmonic under F1, and P0 is the amplitude of a low-frequency nasal peak (~250 Hz) (Chen 1997). In principle, as nasality increases, P0 goes up and A1 is damped, yielding lower A1-P0. However, the details of the relationship between A1 and P0 in natural speech have not been well described. We examined 4778 vowels in French and English elicited words, measuring A1, P0, and the surrounding harmonic amplitudes, and comparing oral and nasal tokens (phonemic nasal vowels in French, and coarticulatorily nasalized vowels in English). Linear mixed-effects regressions confirmed that A1-P0 is predictive of nasality: 4.16 dB lower in English nasal contexts relative to oral and 5.73 dB lower in French (both $p<0.001$). In English, as expected, P0 increased 1.42 dB and A1 decreased 3.93 dB ($p<0.001$). In French, however, both A1 and P0 lowered with nasality (5.73 and 0.93 dB, respectively, $p<0.001$). Even so, in both languages, P0 became more prominent relative to adjacent harmonics in nasal vowels. These data reveal cross-linguistic differences in the acoustic realization of nasal vowels and suggest P0 prominence as a potential perceptual cue to be investigated.

1aSC9. Impact of mismatch conditions between mobile phone recordings on forensic voice comparison. Balamurali B T Nair, Esam A. Alzqhoul, and Bernard J. Guillemin (Dept. of Elec. and Comput., The Univ. of Auckland, Bldg. 303, Rm. 240, Level 2, Sci. Ctr., 38 Princes St., Auckland, Auckland, Auckland 1142, New Zealand, bbah005@aucklanduni.ac.nz)

Mismatched conditions between the recordings of suspect, offender and relevant background population represent a typical scenario in real forensic casework. In this paper, we investigate the impact of mismatch conditions associated with mobile phone speech recordings on forensic voice comparison (FVC). The two major mobile phone technologies currently in use today are the Global System for Mobile Communications (GSM) and Code Division Multiple Access (CDMA). These are fundamentally different in the way in which they handle the speech signal, which in turn will lead to significant mismatch between speech recordings. Our results suggest that the resulting degradation on the accuracy of a FVC analysis can be very significant (as high as 150%). Surprisingly, though, our results also suggest that the reliability of a FVC analysis may actually improve. We propose a strategy for lessening this impact by passing the suspect speech data through the GSM or CDMA codecs, depending on the network origin of the offender data, prior to the FVC analysis. Though this goes a long way to mitigating the impact (a reduction in loss of accuracy from 150% to 80%), it is still not as good as analysis under matched conditions.

1aSC10. 99.8 percent accuracy achieved on Peterson and Barney (1952) acoustic measurements. Michael A. Stokes (R & D, Waveform Commun., 3929 Graceland Ave., Indianapolis, IN 46208, waveform.model@yahoo.com)

In 2012, a paper was presented (Reetz, 2012) discussing the lack of working phonemic models, which was an acknowledgment to an earlier presentation (Ladefoged, 2004) discussing 50+ years of phonetics and phonology. These presentations highlighted the successes in phonological research over the last 60 and 50 years, respectively, but both concluded that there is still no recognized working model of phoneme identification. This presentation will discuss the Waveform Model of Vowel Perception (Stokes, 2009) achieving 99.8% accuracy on the Peterson and Barney (1952) dataset using 30 conditional statements across all ten vowels produced by the 33 males (509/510 for the vowels identified by humans at 100%). These results replicate and improve on the 99.2% achieved across the vowels produced by the males in the Hillenbrand (1995) dataset (Stokes, 2011). As a logical progression, ELBOW was developed in 2013 using the algorithm developed for static data to identify streaming vowel productions achieving over 91% before introducing improvements. Beyond ELBOW, it was essential to replicate earlier results on the most cited dataset in the literature. The Waveform Model has now replicated human performance across multiple datasets and is being successfully introduced into automatic speech recognition applications.

1aSC11. Lombard effect based speech analysis across noisy environments for voice communications with cochlear implant subjects. Jaewook Lee, Hussnain Ali, Ali Ziaei, and Jonh H. Hansen (Elec. Eng., Univ. of Texas at Dallas, 800 West Campbell Rd., EC33, Office ECSN 4.414, Richardson, TX 75080, jaewook@utdallas.edu)

Changes in speech production including vocal effort based on auditory feedback are an important research domain for improved human communication. For example, in the presence of environmental noise, a speaker experiences the well-known phenomenon known as Lombard effect. Lombard effect has been studied for normal hearing listeners as well as for automatic speech/speaker recognition systems, but not for cochlear implant (CI) recipients. The objective of this study is to analyze the speech production of CI users with respect to environmental change. We observe and study this effect using mobile personal audio recordings from continuous single-session audio streams collected over an individual's daily life. Prior advancements in this domain include the "Prof-Life-Log" longitudinal study at UTDallas. Four CI speakers participated by producing read and spontaneous speech in six naturalistic noisy environments (e.g., office, car, outdoor, cafeteria, etc.). A number of speech production parameters (e.g., short-time log-energy, fundamental frequency, etc.) known to be sensitive to Lombard speech were measured for both communicative and non-communicative speech as a function of environment. Results indicate that variability in the speech production parameters were found in the upward direction with an increase in background noise level. Overall higher values in acoustic variables were observed in the inter-personal conversations related to the non-conversational speech.

Session 1aSP

Signal Processing in Acoustics: Sampling Methods for Bayesian Signal Processing

Cameron J. Fackler, Cochair
Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th St, Troy, NY 12180

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180

Invited Papers

8:40

1aSP1. Statistical sampling and Bayesian illumination waveform design for multiple-hypothesis target classification in cognitive signal processing. Grace A. Clark (Grace Clark Signal Sci., 532 Alden Ln., Livermore, CA 94550, clarkga1@comcast.net)

Statistical sampling algorithms are widely used in Bayesian signal processing for drawing real-valued independent, identically distributed (i.i.d.) samples from a desired distribution. This paper focuses on the more difficult problem of how to draw complex correlated samples from a distribution specified by both an arbitrary desired probability density function and a desired power spectral density. This problem arises in cognitive signal processing. A cognitive signal processing system (for example, in radar or sonar) is one that observes and learns from the environment; then uses a dynamic closed-loop feedback mechanism to adapt the illumination waveform so as to provide system performance improvements over traditional systems. Current cognitive radar algorithms focus only on target impulse responses that are Gaussian distributed to achieve mathematical tractability. This research generalizes the cognitive radar target classifier to deal effectively with arbitrary non-Gaussian distributed target responses. The key contribution lies in the use of a kernel density estimator and an extension of a new algorithm by Nichols *et al.* for drawing complex correlated samples from target distributions specified by both an arbitrary desired probability density function and a desired power spectral density. Simulations using non-Gaussian target impulse response waveforms demonstrate very effective classification performance.

9:00

1aSP2. Bayesian inversion and sequential Monte Carlo sampling techniques applied to nearfield acoustic sensor arrays. Mingsian R. Bai (Power Mech. Eng., Tsing Hua Univ., 101 sec.2, Kuang_Fu Rd., Hsinchu 30013, Taiwan, msbai63@gmail.com), Amal Agarwal (Power Mech. Eng., Tsing Hua Univ., Mumbai, India), Ching-Cheng Chen, and Yen-Chih Wang (Power Mech. Eng., Tsing Hua Univ., Taipei, Taiwan)

This paper demonstrates that inverse source reconstruction can be performed using a methodology of particle filters that relies primarily on the Bayesian approach of parameter estimation. The proposed approach is applied in the context of nearfield acoustic holography based on the equivalent source method (ESM). A state-space model is formulated in light of the ESM. The parameters to estimate are amplitudes and locations of the equivalent sources. The parameters constitute the state vector which follows a first-order Markov process with the transition matrix being the identity for every frequency-domain data frame. The implementation of recursive Bayesian filters involves a sequential Monte Carlo sampling procedure that treats the estimates as point masses with a discrete probability mass function (PMF) which evolves with iteration. It is evident from the results that the inclusion of the appropriate prior distribution is crucial in the parameter estimation.

9:20

1aSP3. Bayesian sampling for practical design of multilayer microperforated panel absorbers. Cameron J. Fackler and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St, Greene Bldg., Troy, NY 12180, facklc@rpi.edu)

Bayesian sampling is applied to produce practical designs for microperforated panel acoustic absorbers. Microperforated panels have the capability to produce acoustic absorbers with very high absorption coefficients, without the use of porous materials. However, the absorption produced by a single panel is limited to a narrow frequency range, particularly at high absorption coefficient values. To provide broadband absorption, multiple microperforated panel layers may be combined into a multilayer absorber. To design such an absorber, the necessary number of layers must be determined and four design parameters must be specified for each layer. Using Bayesian model selection and parameter estimation, this work presents a practical method for designing multilayer microperforated panel absorbers. Particular attention is paid to aspects of the underlying sampling method that enable automatic handling of design constraints such as limitations of the manufacturing process and availability of raw materials.

1a MON. AM

9:40

1aSP4. Particle filtering for robust modal identification and sediment sound speed estimation. Nattapol Aunsri and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

Bayesian methods provide a wealth of information on acoustic features of a propagation medium and the uncertainty surrounding their estimation. In previous work, we showed how sequential Bayesian (particle) filtering can be used to extract dispersion characteristics of a waveguide. Here, we utilize these characteristics for the estimation of geoacoustic properties of sediments. As expected, the method relies on accurate identification of modes. The effect of correct/erroneous mode identification on geoacoustic estimates is quantified and approaches are developed for robust modal recognition in conjunction with the particle filter. Additionally, the statistical behavior of the noise present in the data measurements is further investigated with more complex noise modeling leading to improved results. The approaches are validated with both synthetic and real data collected during the Gulf of Mexico Experiment. [Work supported by ONR.]

10:00–10:20 Break

10:20

1aSP5. Efficient trans-dimensional Bayesian inversion for geoacoustic profile estimation. Stan E. Dosso, Jan Dettmer, Gavin Steininger (School of Earth & Ocean Sci, Univ. of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

This paper considers sampling efficiency of trans-dimensional (trans-D) Bayesian inversion based on the reversible-jump Markov-chain Monte Carlo (rjMCMC) algorithm, with application to seabed acoustic reflectivity inversion. Trans-D inversion is applied to sample the posterior probability density over geoacoustic parameters for an unknown number of seabed layers, providing profile estimates with uncertainties that include the uncertainty in the model parameterization. However, the approach is computationally intensive. The efficiency of rjMCMC sampling is largely determined by the proposal schemes applied to perturb existing parameters and to assign values for parameters added to the model. Several proposal schemes are examined, some of which appear new for trans-D geoacoustic inversion. Perturbations of existing parameters are considered in a principal-component space based on an eigen-decomposition of the unit-lag parameter covariance matrix (computed from successive models along the Markov chain, a diminishing adaptation). The relative efficiency of proposing new parameters from the prior versus a Gaussian distribution focused near existing values is considered. Parallel tempering, which employs a sequence of interacting Markov chains with successively relaxed likelihoods, is also considered to increase the acceptance rate of new layers. The relative efficiency of various proposal schemes is compared through repeated inversions with a pragmatic convergence criterion.

10:40

1aSP6. Bayesian tsunami-waveform inversion with trans-dimensional tsunami-source models. Jan Dettmer (Res. School of Earth Sci., Australian National Univ., 3800 Finnerty Rd., Victoria, Br. Columbia V8W 3P6, Canada, jand@uvic.ca), Jakir Hossen, Phil R. Cummins (Res. School of Earth Sci., Australian National Univ., Canberra, ACT, Australia), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper develops a self-parametrized Bayesian inversion to infer the spatio-temporal evolution of tsunami sources (initial sea state) due to megathrust earthquakes. To date, tsunami-source uncertainties are poorly understood, and the effect of choices such as discretization have not been studied. The approach developed here is based on a trans-dimensional self-parametrization of the sea surface, avoids regularization constraints and provides rigorous uncertainty estimation that accounts for model-selection ambiguity associated with the source discretization. The sea surface is parametrized using self-adapting irregular grids, which match the local resolving power of the data and provide parsimonious solutions for complex source characteristics. Source causality is ensured by including rupture-velocity and obtaining delay times from the Eikonal equation. The data are recorded on ocean-bottom pressure and coastal wave gauges and predictions are based on Green-function libraries computed from ocean-basin scale tsunami models for cases that include/exclude dispersion effects. The inversion is applied to tsunami waveforms from the great 2011 Tohoku-Oki (Japan) earthquake. The tsunami source is strongest near the Japan trench with posterior mean amplitudes of ~5 m. In addition, the data appear sensitive to rupture velocity, which is part of our kinematic source model.

Contributed Paper

11:00

1aSP7. Model selection using Bayesian samples: An introduction to the deviance information criterion. Gavin Steininger, Stan E. Dosso, Jan Dettmer (SEOS, U Vic, 201 1026 Johnson St., Victoria, BC v7v 3n7, Canada, gavin.amw.steininger@gmail.com), and Charles W. Holland (SEOS, U Vic, State College, Pennsylvania)

This paper presents the deviance information criterion (DIC) as a metric for model selection based on Bayesian sampling approaches, with examples from seabed geoacoustic and/or scattering inversion. The DIC uses all samples of a distribution to approximate Bayesian evidence, unlike more common measures such as the Bayesian information criterion, which only use point estimates. The DIC uses distribution samples to approximate Bayesian evidence, unlike more common measures such as the Bayesian information criterion based on point estimates. Hence the DIC is more appropriate for non-linear Bayesian inversions utilizing posterior sampling. Two examples are considered: determining the dominant seabed scattering mechanism (interface and/or volume scattering), and choosing between seabed profile parameterizations based on smooth gradients (polynomial splines) or discontinuous homogeneous layers. In both cases, the DIC is applied to trans-dimensional inversions of simulated and measured data, utilizing reversible jump Markov chain Monte Carlo sampling. For the first case, the DIC is found to correctly select the true scattering mechanism for simulations, and its choice for the measured data inversion is consistent with sediment cores extracted at the experimental site. For the second case, the DIC selects the polynomial spline parameterization for soft seabeds with smooth gradients. [Work supported by ONR.]

Session 1aUW

Underwater Acoustics: Understanding the Target/Waveguide System–Measurement and Modeling I

Kevin L. Williams, Chair
Applied Physics Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105

Chair's Introduction—8:45

Invited Papers

8:50

1aUW1. Very-high-speed 3-dimensional modeling of littoral target scattering. David Burnett (Naval Surface Warfare Ctr., Code CD10, 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

NSWC PCD has developed a high-fidelity 3-D finite-element (FE) modeling system that computes acoustic color templates (target strength vs. frequency and aspect angle) of single or multiple realistic objects (e.g., target + clutter) in littoral environments. High-fidelity means that 3-D physics is used in all solids and fluids, including even thin shells, so that solutions include not only all propagating waves but also all evanescent waves, the latter critically affecting the former. Although novel modeling techniques have accelerated the code by several orders of magnitude, it takes about one day to compute an acoustic color template. However, NSWC PCD wants to be able to compute thousands of templates quickly, varying target/environment features by small amounts, in order to develop statistically robust classification algorithms. To accomplish this, NSWC PCD is implementing a radically different FE technology that has already been developed and verified. It preserves all the 3-D physics but promises to accelerate the code another two to three orders of magnitude. Porting the code to an HPC center will accelerate it another one to two orders of magnitude, bringing performance to seconds per template. The talk will briefly review the existing system and then describe the new technology.

9:10

1aUW2. Modeling three-dimensional acoustic scattering from targets near an elastic bottom using an interior-transmission formulation. Saikat Dey, William G. Szymczak (Code 7131, NRL, 4555 Overlook Ave. SW, Washington, DC 20375, saikat.dey@nrl.navy.mil), Angie Sarkissian (Code 7130, NRL, Washington, DC), Joseph Bucaro (Excet Inc., Springfield, VA), and Brian Houston (Code 7130, NRL, Washington, DC)

For targets near the sediment–fluid interface, the scattering response is fundamentally influenced by the characterization of the sediment in the model. We show that if the model consists of a three-dimensional elastic sediment with acoustic fluid on top, then the use of perfectly matched-layer (PML) approximation for the truncation of the infinite exterior domain for scattering applications has fundamental problems and gives erroneous results. We present a novel formulation using the an interior-transmission representation of the scattering problem where the exterior truncation with PML does not induce errors in the result. Numerical examples will be presented to verify the application of this formulation to scattering from elastic targets near a fluid–sediment interface.

9:30

1aUW3. The fluid–structure interaction technique specialized to axially symmetric targets. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com) and Petr Krysl (Structural Eng., Univ. of California, San Diego, La Jolla, CA)

The fluid–structure interaction technique provides a paradigm for solving scattering from elastic targets embedded in a fluid by a combination of finite and boundary element methods. In this technique, the finite element method is used to compute the target's impedance matrix and the Helmholtz–Kirchhoff integral with the appropriate Green's function is used to represent the field in the exterior medium. The two equations are coupled at the surface of the target by imposing the continuity of pressure and normal displacement. This results in a Helmholtz–Kirchhoff boundary element equation that can be used to compute the scattered field anywhere in the surrounding environment. This method reduces a finite element problem to a boundary element one with drastic reduction in the number of unknowns, which translates to a significant reduction in numerical cost. This method was developed and tested for general 3D targets. In this paper, the method is specialized to axially symmetric targets, which provides further reduction in numerical cost, and validated using benchmark solutions.

Contributed Papers

9:50

1aUW4. A new T matrix for acoustic target scattering by elongated objects in free-field and in bounded environments. Raymond Lim (Code X11, NSWC Panama City Div., 110 Vernon Ave., Code X11, Panama City, FL 32407-7001, raymond.lim@navy.mil)

The transition (T) matrix of Waterman has been very useful for computing fast, accurate acoustic scattering predictions for axisymmetric elastic objects but this technique is usually limited to fairly smooth objects that are not too aspherical unless complex basis functions or stabilization schemes are used. To remove this difficulty, a spherical-basis formulation adapted from approaches proposed recently by Waterman [J. Acoust. Soc. Am. 125, 42–51 (2009)] and Doicu, *et al*. [*Acoustic & Electromagnetic Scattering Analysis Using Discrete Sources*, Academic Press, London, 2000] is suggested. The new method is implemented by simply transforming the high-order outgoing spherical basis functions within standard T-matrix formulations to low-order functions distributed along the object's symmetry axis. A free-field T-matrix is produced in a nonstandard form but computations with it become much more stable for aspherical shapes. Some advantages of this approach over Waterman's and Doicu, *et al*.'s approaches are noted and, despite its nonstandard form, the feasibility of extension to objects in a plane-stratified environment is demonstrated. Sample calculations for an elongated spheroid demonstrate the enhanced stability.

10:05–10:20 Break

10:20

1aUW5. Kirchhoff approximation for spheres and cylinders partially exposed at flat surfaces and application to the interpretation of backscattering. Aaron M. Gunderson, Anthony R. Smith, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, aaron.gunderson01@gmail.com)

For cylinders partially exposed at flat surfaces, the Kirchhoff approximation was previously evaluated analytically and compared with measured backscattering at a free surface as a function of exposure [K. Baik and P. L. Marston, IEEE J. Ocean. Eng. 33, 386–396 (2008)]. In the present research, this approach is extended to the cases of numerical integration for high frequency backscattering by partially exposed spheres and cylinders. The cylinder case was limited to broadside illumination at grazing incidence for which one-dimensional integration is sufficient and the limits of integration were previously discussed by Baik and Marston. In the corresponding sphere case, however, two-dimensional integration is required and the corresponding limits of integration become complicated functions of the amount of exposure and the grazing angle of the illumination. These approximations of the backscattering, while they omit Franz wave and elastic contributions, are useful for modeling the evolution of how the reflected scattering contributions depend on the target exposure. They are also useful for understanding the time evolution of specular scattering contributions. The sphere case was compared with the exact analysis of backscattering by a half exposed rigid sphere at a free surface that also displays partially reflected Franz wave contributions. [Work supported by ONR.]

Invited Papers

10:35

1aUW6. Acoustic ray model for the scattering from an object on the sea floor. Steven G. Kargl, Aubrey L. Espana, and Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, kargl@uw.edu)

Target scattering within a waveguide is recast into a ray model, where time-of-flight wave packets are tracked. The waveguide is replaced by an equivalent set of image sources and receivers, where rays are associated with these images and interactions with the waveguide's boundaries are taken into account. By transforming wave packets into the frequency domain, scattering becomes a multiplication of a wave packet's spectrum at the target location and the target's free-field scattering amplitude. Data- and model-model comparisons for an aluminum replica of a 100-mm unexploded ordnance will be discussed. For the data-model comparisons, synthetic aperture sonar (SAS) data were collect during Pond Experiment 2010 from this replica, where it was placed on a water-sand sediment boundary. The model-model comparisons use the results from a hybrid 2-D/3-D model. The hybrid model combines a 2D finite-element model to predict the scattered pressure and its derivatives in the near-field of the target, and then a 3D Helmholtz integral to propagate the pressure to the far field. The data- and model-model comparisons demonstrate the viability of using the ray model to quickly generate realistic pings suitable for both SAS and acoustic color template processing. [Research supported by SERDP and ONR.]

10:55

1aUW7. Orientation dependence for backscattering from a solid cylinder near an interface: Imaging and spectral properties. Daniel Plotnick, Philip L. Marston (Washington State Univ., 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com), Aubrey Espana, and Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

When a solid cylinder lies proud on horizontal sand sediment significant contributions to backscattering, specular and elastic, involve multipath reflections from the cylinder and interface. The scattering structure and resulting spectrum versus azimuthal angle, the "acoustic template," may be understood using a geometric model [K. L. Williams *et al*., J. Acoust. Soc. Am. 127, 3356–3371 (2010)]. If the cylinder is tilted such that the cylinder axis is no longer parallel to the interface the multipath structure is modified. Some changes in the acoustic template can be approximately modeled using a combination of geometric and physical acoustics. For near broadside scattering the analysis gives a simple expression relating certain changes in the template to the orientation of the cylinder and the source geometry. These changes are useful for inferring the cylinder orientation from the scattering. Changes to the template at end-on and intermediate angles are also examined. The resulting acoustic images show strong dependence on the cylinder orientation in agreement with this model. A similar model applies to a metallic cylinder adjacent to a flat free surface and was confirmed in tank experiments. The effect of vertical tilt on the acoustic image was also investigated. [Work supported by ONR.]

11:15

1aUW8. Acoustic scattering enhancements for partially exposed cylinders in sand and at a free surface caused by Franz waves and other processes. Anthony R. Smith, Aaron M. Gunderson, Daniel S. Plotnick, Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA, spacetime82@gmail.com), and Grant C. Eastland (NW Fisheries Sci. Ctr., Frank Orth & Assoc. (NOAA Affiliate), Seattle, WA)

Creeping waves on solid cylinders having slightly subsonic phase velocities and large radiation damping are described as Franz waves because of association with complex poles investigated by Franz. For free-field high frequency broadside backscattering in water, the associated echoes are weak due to radiation damping. It was recently demonstrated, however, that for partially exposed solid metal cylinders at a free surface viewed at grazing incidence, the Franz wave echo can be large relative to the specular echo when the grazing angle is sufficiently small [G. C. Eastland and P. L. Marston, J. Acoust. Soc. Am. 135, 2489–2492 (2014)]. The Fresnel zone associated with the specular echo is occluded making it weak while the Franz wave is partially reflected at the interface behind the cylinder. This hypothesis is also supported by calculating the exact backscattering by half-exposed infinitely long rigid cylinders viewed over a range of grazing angles. Additional experiments concern the high frequency backscattering by cylinders partially buried in sand viewed at small grazing angles. From the time evolution of the associated backscattering by short tone bursts, situations have been identified for which partially reflected Franz wave contributions become significant. Franz waves may contribute to sonar clutter from rocks. [Work supported by ONR.]

11:35

1aUW9. Pressure gradient coupling to an asymmetric cylinder at an interface. Christopher Dudley (NSWC PCD, 110 Vernon Ave., Panama City, FL 32407, mhhd@hotmail.com)

Invited Abstract Special session: “Investigation of target response near interfaces, where coupling between target and environmental properties are important.” Acoustic scattering results from solid and hollow notched aluminum cylinders are presented as a function of the incident angle. This flat machined into the circular cylinder resembles the topography(geometry) of an finned unexploded ordnance (UXO). Prior experiments have shown selective coupling to modes of a flat ended cylinder and the effect of pressure nodes to coupling to a similar notched cylinder [Espana et al., J. Acoust. Soc. Am. 126, 2187 (2009) and Marston & Marston, J. Acoust. Soc. Am. 127, 1750 (2010)]. The wavefront crossing the flat face of the notch in the paddle has a pressure gradient when not co-linear with the normal to the flat face of the notch. This pressure gradient applies a torque to the cylinder. Torsional modes can be setup in multiple scaled version of the pseudo-UXOs. Analysis of scattering experiments in the Gulf of Mexico and laboratory scale water tanks indicate robust returns form these fin like targets.

MONDAY AFTERNOON, 27 OCTOBER 2014 MARRIOTT 7/8, 1:00 P.M. TO 5:15 P.M.

Session 1pAA

Architectural Acoustics: Computer Auralization as an Aid to Acoustically Proper Owner/Architect Design Decisions

Robert C. Coffeen, Cochair
Architecture, University of Kansas, 4721 Balmoral Drive, Lawrence, KS 66047

Kevin Butler, Cochair
Henderson Engineers, Inc., 8345 Lenexa Dr., #300, Lenexa, KS 66214

Chair’s Introduction—1:00

Invited Papers

1:05

1pAA1. The impact of auralization on design decisions for the House of Commons of the Canadian Parliament. Ronald Eligator (Acoustic Distinctions, 145 Huguenot St., New Rochelle, NY 10801, religator@ad-ny.com)

The House of Commons of the Canadian Parliament will be temporary relocated to a 27,000 m^3 glass-enclosed atrium with stone and glass walls while their home Chamber is being renovated and restored. Acoustic goals include excellent speech intelligibility for Members and guests in the room, and production of high-quality audio recordings of all proceedings for live and recorded streaming and broadcast. Room modeling and auralization using CATT Acoustic has been used to evaluate the acoustic environment of the temporary

location during design. Modeling and testing of the current House Chamber has also been performed to validate the results and conclusions drawn from the model of the new space. The use of auralizations has helped the Owner and Architect understand the impact of design choices on the achievement of the acoustic performance goals, and smoothed the path for the integration of design features that might otherwise have been difficult for them to accept. Measured and calculated data as well as audio examples will be presented.

1:25

1pAA2. Cost effective auralizations to help architects and owners make informed decisions for sound isolating assemblies. David Manley and Ben Bridgewater (D.L. Adams Assoc., Inc., 1536 Ogden St., Denver, CO 80218, dmanley@dlaa.com)

As an acoustical consultant, subjective descriptions of noise environments only get you so far. For example, it can be difficult for an Architect to qualify the difference between STC 35 and STC 40 windows on a given office space next to a highway. Often, justifying the increased cost for the increased sound isolation performance is at the forefront of the decision making process for the Owner and Architect. To help them understand the relative difference in performance, DLAA uses a simplified auralization process to create audio demonstrations of the difference between sound isolating assemblies. This presentation will discuss the process of creating the auralizations and review case studies where the auralizations helped the client make a more informed decision.

1:45

1pAA3. Using auralization to aid in decision making to meet customer requirements for room response and speech intelligibility. Thomas Tyson (Professional Systems Div., Bose, 5160 South Deborah Ct., Springfield, MO 65810, Tom_Tyson@bose.com)

To meet specific design goals such as a high degree of speech intelligibility along with targeted reverberation time; the presenter will show how the use of auralization can help determine the effectiveness of acoustic treatments and loudspeaker directivity types, beyond just the use of predicted numerical data.

2:05

1pAA4. Bridging the gap between eyes and ears with auralization. Robin S. Glosemeyer Petrone, Scott D. Pfeiffer (Threshold Acoust..com, 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, robin@thresholdacoustics.com), and Marcus Mayell (Judson Univ., Elgin, IL)

Ray trace animation, level plots, and impulse responses, while all useful tools in providing a visual representation of sound, do not always bridge the gap between the eye and ear. Threshold utilized auralization to inform decisions for an upcoming theater renovation with the goal of improving the room's acoustic support of orchestral performance. To achieve the desired acoustic response, the renovation will require major modifications to the shaping of a hall with a very distinctive architectural vernacular; a distinctive vernacular that will need to be preserved in some form to maintain the facility's identity. Along with other modeling tools, auralization provided useful support, reassuring both the client and the design team of the validity of the concepts.

2:25

1pAA5. Extended tools for simulated impulse responses. Wolfgang Ahnert and Stefan Feistel (Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

To calculate impulse responses is already done since more than 25 years. The routines did allow simple calculations without and now always with scattered sound components. Today, sophistic routines calculate frequency-dependent full impulse responses comparable with measured ones. Parallel to this development, auralization routines have been developed first for monaural and binaural reproduction and nowadays ambisonic signals are created in B-Format of first and second order. These signals make audible during the reproduction in an ambisonic playback configuration the distribution of wall and ceiling reflections in computer models in EASE. Beside the acoustic detection of desired or unwanted reflections, which always is asking for the correct reproduction of the ambisonic signals the visualization of the reflection distribution is desired. In EASE, a new tool has been implemented to correlate the reflections in an impulse response with their position in a 3D presentation. This new hedgehog presentation of full impulse responses correlates angle-dependent with the view position of the model. So, any wanted or unwanted reflections may be identified quickly. A comparison with ambisonic signals via auralization is possible.

2:45

1pAA6. Auralization as an aid in decision-making: Examples from professional practice. Benjamin Markham, Robert Connick, and Jonah Sacks (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

The authors and our colleagues have presented dozens of auralizations in the service of our architectural acoustics consulting work, on projects ranging from large atriums to classrooms to sound isolation between nightclubs and surrounding facilities (and many others). The aim of most of these presentations is to communicate the relative efficacy of design alternatives or acoustical treatment options. In some cases, the effects are profound; in others, the acoustical impact may be rather subtle. Without perfect correlation, we have noted a general trend: when the observable change in acoustical attributes presented in the auralization is substantial, so too is the interest on the part of the owner to invest in significant or even aggressive acoustical design alternatives; by contrast, subtler changes in perceived acoustical character often leave owners and architects less inclined to dedicate design resources to pursue alternatives that differ from the architect or owner's original vision. Examples of auralizations following (and contradicting) this trend will be presented, along with descriptions of the design direction taken following meetings and discussions that accompanied the auralizations.

3:05–3:20 Break

3:20

1pAA7. Auralization and the real world. Shane J. Kanter (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com), Ben Bridgewater (D.L. Adams, Denver, CO), and Robert C. Coffeen (School of Architecture, Design & Planning, The Univ. of Kansas, Lawrence, KS)

Architects value their senses and strive to design spaces that are engaging all five of them. However, architects typically make design decisions based primarily on how spaces appear and feel, as opposed to acousticians who normally justify design intent with the use of numbers, graphs, and charts. Although the data are clear to acousticians, auralizations are a useful tool to engage architects, building owners, and other clients and their sense of hearing to help them make informed decisions. If auralizations are used to demonstrate the effect of design decisions based on acoustics, there must be confidence in the accuracy and realism of these audio simulations. In order to better understand the accuracy and realism of auralizations, a study was conducted comparing auralizations created from models of an existing facility to listening within the facility. Listeners were asked to compare the "real world" sound to the auralizations of this sound by completing a survey with questions focusing on such comparisons. By presenting the actual sound and the auralizations in the same space, a direct comparison can be made and the accuracy and realism of the auralizations can be determined. Results and observations from the study will be presented.

3:40

1pAA8. Directing room acoustic decisions for a college auditorium renovation by using auralization. Robert C. Coffeen (Architecture, Univ. of Kansas, 4721 Balmoral Dr., Lawrence, KS 66047, rcoffeen@ku.edu)

From an acoustical viewpoint, the renovation of a multipurpose college auditorium was predicted by music and theater faculty to be a compromise not suitable for either music or theater. It was obvious that either variable sound absorption or active acoustics would be required to satisfy the multipurpose uses of the auditorium. Active acoustics was rejected by the college due to cost and an experience by one faculty member. And the faculty committee was not familiar with variable sound absorption. Using a computer model of the auditorium it was determined that the volume of the venue could be established to produce the desired maximum reverberation time for music and that vertical rising drapery could produce the desired reverberation time for drama. Auralization was used to demonstrate to the faculty committee that with variable sound absorption the auditorium could properly accommodate music of various types and theatrical performances including drama.

Contributed Papers

4:00

1pAA9. "Illuminating" reflection orders in architectural acoustics using SketchUp and light rendering. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

The conventional architecture workflow tends to—quite literally—"overlook" matters of sound, given that the modeling tools of architectural design are almost exclusively visual in nature. The modeling tools used by architectural acousticians, however, produce visual representations, which are, frankly, less than inspirational for the design process. This project develops a simple scheme to visualize acoustic reflection orders using light rendering in the freely available and widely used Trimble SketchUp 3D modeling software. In addition to allowing architectural designers to visualize acoustic reflections in a familiar modeling environment, this scheme also works easily with complex geometry. The technique and examples will be presented.

4:15

1pAA10. Using auralization to evaluate the decay characteristics that impact intelligibility in a school auditorium. Bruce C. Olson (Ahnert Feistel Media Group, 8717 Humboldt Ave. North, Brooklyn Park, MN 55444, bcolson@afmg.eu) and Bruce C. Olson (Olson Sound Design, Brooklyn Park, MN)

Auralization was used to evaluate the effectiveness of the loudspeaker design in a high school auditorium to provide good speech intelligibility when used for lectures. The goals of this project where to offer an aural impression that enhances the visual printouts of the simulation results from the 3D model of the space in EASE using the Analysis Utility for Room Acoustics. The process used will be described and some of the results will be presented.

4:30

1pAA11. Vibrolization: Simulating whole-body structural vibration for clients and colleagues with the Motion Platform. Clemeth Abercrombie (Acoust., Arup, New York, NY), Tom Wilcock (Adv. Tech. and Res., Arup, New York, NY), and Andrew Morgan (Acoust., Arup, 77 Water St., Arup, New York, NY 10005, andrew.morgan@arup.com)

Arup has recently introduced an experiential design tool for demonstrating whole-body vibration. The Motion Platform, a bespoke simulator, moves vertically and can reproduce structural vibration in buildings, transport, and any other situations that involve shaking. Beyond humans, the platform can also shake objects—opening the door for developing new vibration criteria for devices such as video cameras and projectors. We will share our experience in developing the platform and how it has helped us communicate design ideas to clients and design team members.

4:45

1pAA12. The role of auralization utilizing the end user source signal in determining final material finishes for the Chapel at St. Dominics. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

The Chapel at St. Dominics Hospital in Jackson, Mississippi, was created for religious services, prayer time, and serve other spiritual needs of the hospital's patients, employees, medical staff, hospital visitors, and the greater community. It is an intimate space seating up to 100 people and is used daily by the Dominican Sisters, who first started the Jackson Infirmary in 1946. This paper outlines the process used to record the voices of the sisters and then use them to generate auralizations, which helped drive decisions regarding acoustic finishes.

5:00

1pAA13. The construction and implementation of a multichannel loudspeaker array for accurate spatial reproduction of sound fields. Matthew T. Neal, Colton D. Snell, and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

The spatial distribution of sound has a strong impact upon a listener's overall impression of a room and must be reproduced accurately for auralization. In concert hall acoustics, directionally independent metrics such as reverberation time and clarity index simply do not predict this impression. Late lateral energy level, lateral energy fraction, and the interaural correlation coefficient are measures of spatial impression, but more work is needed before we fully understand how the directional distribution of sound should influence architectural design decisions. A three-dimensional array of 28 loudspeakers and two subwoofers has been constructed in a hemi-anechoic chamber at PSU, allowing for accurate reproduction of sound fields. For the array, closed-box loudspeakers were built and digitally equalized to ensure a flat frequency response. With this facility, subjective studies investigating spatial sound in concert halls can be conducted using measured sound fields and perceptually motivated auralizations, not tied to a physical room. Such a facility is instrumental in understanding and communicating subtle differences in sound fields to listeners, whether they be musicians, architects, or clients. The flexibility and versatility of this system will facilitate room acoustics research at Penn State for years to come. [Work supported by NSF Award 1302741.]

MONDAY AFTERNOON, 27 OCTOBER 2014 LINCOLN, 1:00 P.M. TO 5:00 P.M.

Session 1pAB

Animal Bioacoustics and Signal Processing in Acoustics: Array Localization of Vocalizing Animals

Michelle Fournet, Cochair
College or Earth Ocean and Atmospheric Sciences, Oregon State University, 425 SE Bridgeway Ave., Corvallis, OR 97333

David K. Mellinger, Cochair
Coop. Inst. for Marine Resources Studies, Oregon State University, 2030 SE Marine Science Dr., Newport, OR 97365

Chair's Introduction—1:00

Invited Papers

1:05

1pAB1. Exploiting the sound-speed minimum to extend tracking ranges of vertical arrays in deep water environments. Aaron Thode, Delphine Mathias (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Janice Straley (Univ. of Alaska, Southeast, Sitka, AK), Russel D. Andrews (Alaska SeaLife Ctr., Seward, AK), Chris Lunsford, John Moran (Auke Bay Labs., NOAA, Juneau, AK), Jit Sarkar, Chris Verlinden, William Hodgkiss, and William Kuperman (SIO, UCSD, La Jolla, CA)

Underwater acoustic vertical arrays can localize sounds by measuring the vertical elevation angles of various multipath arrivals generated by reflections from the ocean surface and bottom. This information, along with measurements of the relative arrival times of the multipath, can be sufficient for obtaining the range and depth of an acoustic source. At ranges beyond a few kilometers ray refraction effects add additional multipath possibilities; in particular, the existence of a sound-speed minimum in deeper waters permits purely refracted ray arrivals to be detected and distinguished on an array, greatly extending the tracking range for short-aperture systems. Here, two experimental vertical array deployments are presented. The first is a simple two-element system, deployed using longline fishing gear off Sitka, AK. By tracking a tagged sperm whale, this system demonstrated an ability to localize this species out to 35 km range, and provide estimates of the detection range of these animals as a function of sea state. The second deployment—a field trial of an 128-element, mid-frequency vertical array system off Southern California—illustrates how multi-element array gain can further extend the detection and tracking ranges of sperm and humpback whales in deep-water environments. [Work supported by NPRB, NOAA, and ONR.]

1:25

1pAB2. Arrayvolution—An overview of array systems to study bats and toothed whales. Jens C. Koblitz (German Oceanographic Museum, Katharinenberg 14-20, Stralsund 18439, Germany, Jens.Koblitz@meeresmuseum.de), Magnus Wahlberg (Dept. of Biology, RMIT Univ., Odense, Denmark), Peter Stilz (Freelance Biologist, Hechingen, Germany), Jamie MacAulay (Sea Mammal Res. Unit, Univ. of St Andrews, St. Andrews, United Kingdom), Simone Götze, Anna-Maria Seibert (Animal Physiol., Inst. for Neurobiology, Univ. of Tübingen, Tübingen, Germany), Kristin Laidre (Polar Sci. Ctr., Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Hans-Ulrich Schnitzler (Animal Physiol., Inst. for Neurobiology, Univ. of Tübingen, Zübingen, Germany), and Harald Benke (German Oceanographic Museum, Stralsund, Germany)

Some echolocation signal parameters can be studied using a single receiver. However, studying parameters such as source level, directionality, and direction of signal emission require the use of multi-receiver arrays. Acoustic localization allows for determination of the position of echolocators at the time of signal emission, and when multiple animals are present, calls can be assigned to individuals

based on their location. This combination makes large multi-receiver arrays a powerful tool. Here we present an overview of different array configurations used to study both toothed whales and bats, using a suite of systems ranging from semi-3D-minimum receiver number-number-arrays (3D-MINNAs), linear-2-D-over determined arrays (2D-ODAs), to 3-D-over-determined-arrays (3D-ODAs). We discuss approaches to process and summarize the usually large amounts of data. In some studies, the absolute position of an echolocator and not only relative to the array is crucial. Combining acoustic localizations from a source with geo-referenced receivers allows for determining geo-referenced movements of an echolocator. Combining these animal tracks with other geo-referenced data such as hydrographic parameters will allow new insights into habitat use.

1:45

1pAB3. Tracking Cuvier's beaked whales using small aperture arrays. Martin Gassmann, Sean M. Wiggins, and John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9152 Regents Rd., Apt. L, La Jolla, CA 92037, mgassmann@ucsd.edu)

Cuvier beaked whales are deep-diving animals that produce strongly directional sounds using high frequencies (>30 kHz) at which attenuation due to absorption and scattering is high (>8 dB/km). This makes it difficult to track beaked whales in three dimensions with standard large-aperture hydrophone arrays. By embedding two volumetric small-aperture (~1 m element spacing) arrays into a large-aperture (~1 km element spacing) array of five nodes, individuals and even groups of Cuvier beaked whales were tracked in three dimensions continuously up to one hour within an area of 10 km^2 in the Southern California Bight. This passive acoustic tracking technique provides a tool to study the characteristics of beaked whale echolocation, and their behavior during deep-diving.

2:05

1pAB4. Using ocean bottom seismometer networks to better understand fin whale distributions at different spatial scales. Michelle Weirathmueller, William SD Wilcock, and Dax C. Soule (Univ. of Washington, 1503 NE Boat St., Seattle, WA 98105, michw@uw.edu)

Ocean bottom seismometers (OBSs) are designed to monitor ground motion caused by earthquakes, but they also record low frequency vocalizations of fin and blue whales. Seismic networks used for opportunistic whale datasets are rarely optimized for acoustic localization of marine mammals. We demonstrate the use of OBSs for studying fin whales using two different networks. The first example is a small, closely spaced network of 8 OBSs deployed on the Juan de Fuca Ridge from 2003 to 2006. An automated method for identifying arrival times and locating fin whale calls using a grid search was applied to obtain 154 individual fin whale tracks over one year, revealing information on swimming patterns and spatial distribution in the vicinity of a mid ocean ridge. The second example is a network with widely spaced OBSs, such that a given call can only be detected on one instrument. The Cascadia Initiative Experiment is a sparse array of 70 OBSs covering the Juan de Fuca Plate from 2011 to 2015. Localization methods based on differential arrival times are not possible but techniques to locate the range and bearing to fin whales with a single OBS can be applied to constrain larger scale spatial distributions by comparing call densities in different regions.

2:25

1pAB5. Baleen whale localization using hydrophone streamers during seismic reflection surveys. Shima H. Abadi (Lamont–Doherty Earth Observatory, Columbia Univ., 122 Marine Sci. Bldg., University of Washington 1501 NE Boat St., Seattle, Washington 98195, shimah@ldeo.columbia.edu), Maya Tolstoy (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY), William S. D. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA), Timothy J. Crone, and Suzanne M. Carbotte (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY)

Seismic reflection surveys use acoustic energy to image the structure beneath the seafloor, but concern has been raised about their potential impact on marine animals. Most of the energy from seismic surveys is low frequency, so the concern about their impact is focused on Baleen whales that communicate in the same frequency range. To better mitigate against this impact, safety radii are established based on the criteria defined by the National Marine Fisheries Service. Marine mammal observers use visual and acoustic techniques to monitor safety radii during each experiment. However, additional acoustic monitoring, in particular, locating marine mammals, could demonstrate the effectiveness of the observations, and help us understand animal responses to seismic experiments. A novel sound source localization technique using a seismic streamer has been developed. Data from seismic reflection surveys conducted with the R/V Langseth are being analyzed with this method to locate baleen whales and verify the accuracy of visual detections during experiments. The streamer is 8 km long with 636 hydrophones sampled at 500 Hz. The work focuses on time intervals when only a mitigation gun is firing because of marine mammal sightings. [Sponsored by NSF.]

2:45

1pAB6. Faster than real-time automated acoustic localization and call association for humpback whales on the Navy's Pacific Missile Range Facility. Tyler A. Helble (SSC-PAC, 2622 Lincoln Ave., San Diego, CA 92104, tyler.helble@gmail.com), Glenn Ierley, Gerald D'Spain (Scripps Inst. of Oceanogr., San Diego, CA), and Stephen Martin (SSC-PAC, San Diego, CA)

Optimal time difference of arrival (TDOA) methods for acoustically localizing multiple marine mammals have been applied to the data from the Navy's Pacific Missile Range Facility in order to localize and track humpback whales. Modifications to established methods were necessary in order to simultaneously track multiple animals on the range without the need for post-processing and in a fully automated way, while minimizing the number of incorrect localizations. The resulting algorithms were run with no human intervention at computational speeds faster than the data recording speed on over 40 days of acoustic recordings from the range, spanning several years and multiple seasons. Spatial localizations based on correlating sequences of units originating from within the range produce estimates having a standard deviation typically 10 m or less (due primarily to TDOA measurement errors), and a bias of 20 m or less (due to sound speed mismatch). Acoustic modeling and Monte Carlo simulations play a crucial role in minimizing both the variance and bias of TDOA localization methods. These modeling and simulation techniques will be discussed for optimizing array design, and for maximizing the quality of localizations from existing data sets.

3:05

1pAB7. Applications of an adaptive back-propagation method for passive acoustic localizations of marine mammal sounds. Ying-Tsong Lin, Arthur E. Newhall, and James F. Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

An adaptive back-propagation localization method utilizing the dispersion relation of the acoustic modes of low-frequency sound signals is reviewed in this talk. This method employs an adaptive array processing technique (the maximum a posteriori mode filter) to extract the acoustic modes of sound signals, and it is capable of separating signals from noisy data. The concept of the localization algorithm is to back-propagate modes to a location where the modes align with each other. Gauss-Markov inverse theory is applied to make the normal mode back-propagator adaptive to the signal-to-noise ratio (SNR). When the SNR is high, the localization procedure will push the algorithm to achieve high resolution. On the other hand, when the SNR is low, the procedure will try to retain its robustness and reduce the noise effects. Examples will be shown in the talk to demonstrate the localization performance with comparisons to other methods. Applications to baleen whale sounds collected in Cape Cod Bay, Massachusetts, will also be presented. Lastly, population density estimation using this passive acoustic localization method will be discussed.

3:20–3:45 Break

3:45

1pAB8. Tracking porpoise underwater movements in tidal rapids using drifting hydrophone arrays. Jamie D. Macaulay, Doug Gillespie, Simon Northridge, and Jonathan Gordon (SMRU, Univ. of St Andrews, 15 Crichton St., Anstruther, Fife KY103DE, United Kingdom, jdjm@st-andrews.ac.uk)

The growing interest in generating electrical power from tidal currents using tidal turbine generators raises a number of environmental concerns, including the risk that cetaceans might be injured or killed through collision with rotating turbine blades. To understand this risk we need better information on how cetaceans use tidal rapid habitats and in particular their underwater movements and dive behavior. Focusing on harbor porpoises, a European protected species, we have developed an approach which uses time of arrival differences of narrow band high frequency (NBHF) clicks detected on large aperture hydrophone arrays drifting in tidal rapids, to determine dive tracks of porpoises underwater. Probabilistic localization algorithms have been developed to filter echoes and provide accurate 2D or geo-referenced 3D locations. Calibration trials have been carried out that show that the system can provide depth and location data with submeter errors. Data collected over three seasons in tidal races around Scotland has provided new insights into how harbor porpoises are using these unique habitats, information vital for assessing the risk tidal turbines may pose.

4:00

1pAB9. Using a coherent hydrophone array for observing sperm whale range, classification, and shallow-water dive profiles. Duong D. Tran, Wei Huang, Alexander C. Bohn, Delin Wang (Elec. and Comput. Eng., Northeastern Univ., 006 Hayden Hall, 370 Huntington Ave., Boston, MA 02115, wang.del@husky.neu.edu), Zheng Gong, Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Sperm whales in the New England continental shelf and slope were passively localized, in both range and bearing, and classified using a single low-frequency (<2500 Hz), densely sampled, towed horizontal coherent hydrophone array system. Whale bearings were estimated using time-domain beamforming that provided high coherent array gain in sperm whale click signal-to-noise ratio. Whale ranges from the receiver array center were estimated using the moving array triangulation technique from a sequence of whale bearing measurements. Multiple concurrently vocalizing sperm whales, in the far-field of the horizontal receiver array, were distinguished and classified based on their horizontal spatial locations and the inter-pulse intervals of their vocalized click signals. The dive profile was estimated for a sperm whale in the shallow waters of the Gulf of Maine with 160 m water-column depth located close to the array's near-field where depth estimation was feasible by employing time difference of arrival of the direct and multiply reflected click signals received on the horizontal array. By accounting for transmission loss modeled using an ocean waveguide-acoustic propagation model, the sperm whale detection range was found to exceed 60 km in low to moderate sea state conditions after coherent array processing.

4:15

1pAB10. Testing the beam focusing hypothesis in a false killer whale using hydrophone arrays. Laura N. Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laura_kloepper@brown.edu), Paul E. Nachtigall, Adam B. Smith (Zoology, Univ. of Hawaii, Honolulu, HI), John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA), and Jason E. Gaudette (Neurosci., Brown Univ., Providence, RI)

The odontocete sound production system is complex and composed of tissues, air sacs, and a fatty melon. Previous studies suggested that the emitted sonar beam might be actively focused, narrowing depending on target distance. In this study, we further tested this beam focusing hypothesis in a false killer whale (*Pseudorca crassidens*) in a laboratory setting. Using three linear arrays, we recorded the same emitted click at 2, 4, and 7 m distance while the animal performed a target detection task with the target distance varying between 2, 4, and 7 m. For each click, we calculated the beamwidth, intensity, center frequency, and bandwidth as recorded on each array. As the distance from the whale to the array increased, the received click intensity was higher than predicted by spreading loss. Moreover, the beamwidth varied with range as predicted by the focusing model and contrary to a piston model or spherical spreading. These results support the hypothesis that the false killer whale adaptively focuses its sonar beam according to target range. [Work supported by ONR and NSF.]

4:30

1pAB11. Sei whale localization and tracking using a moored, combined horizontal and vertical line array near the New Jersey continental shelf. Arthur E. Newhall, Ying-Tsong Lin, James F. Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 210 Bigelow Lab. MS11, Woods Hole, MA 02543, anewhall@whoi.edu), and Mark F. Baumgartner (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA)

In 2006, a multidisciplinary experiment was conducted in the Mid-Atlantic continental shelf off the New Jersey coast. During a 2 day period in mid-September 2006, more than 200, unconfirmed but identifiable, sei whale (*Balaenoptera borealis*) calls were collected on a moored, combined horizontal and vertical line hydrophone array. Sei whale movements were tracked over long distances (up to tens of kilometers) using a normal mode back propagation method. This approach uses low-frequency, broadband passive sei whale call receptions from a single-station, two-dimensional hydrophone array to perform long distance localization and tracking by exploiting the dispersive nature of propagating acoustic modes in a shallow water environment. Source depth information and the source signal can also be determined from the localization application. This passive whale tracking, combined with the intensive oceanography measurements performed during the experiment, was also used to examine sei whale movements in relation to oceanographic features observed in this region.

4:45

1pAB12. Obtaining underwater acoustic impulse responses via blind channel estimation. Brendan P. Rideout, Eva-Marie Nosal (Dept. of Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, bprideou@hawaii.edu), and Anders Høst-Madsen (Dept. of Elec. Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

Blind channel estimation is the process of obtaining the impulse responses between a source and multiple (arbitrarily placed) receivers without prior knowledge about the source characteristics or the environment. This approach could simplify localization of non-impulsive submerged sound sources (e.g., pinnipeds or cetaceans); the process of picking arrivals (direct and reflected) could be carried out on the estimated impulse responses rather than on the recorded waveforms, thus facilitating the use of time of arrival-based localization approaches. Blind channel estimation could also be useful in estimating the original source signal of a vocalizing animal through deconvolution of the estimated channel impulse responses and the recorded waveforms. In this paper, simulation and controlled pool studies will be used to explore requirements on source and environment characteristics and to quantify blind channel estimation performance for underwater passive acoustic applications.

MONDAY AFTERNOON, 27 OCTOBER 2014 — INDIANA A/B, 1:15 P.M. TO 5:30 P.M.

Session 1pBA

Biomedical Acoustics: Medical Ultrasound

Robert McGough, Chair
Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824

Contributed Papers

1:15

1pBA1. Investigation of fabricated 1 MHz lithium niobate transfer standard ultrasonic transducer. Patchariya Petchpong (Acoust. and Vib. Dept., National Inst. Metrology of Thailand, 75/7 Rama VI Rd., Thungphayathai, Rajthevi, Bangkok 10400, Thailand, patchariya@nimt.or.th) and Yong Tae Kim (Div. of Convergence Technol., Korea Res. Inst. of Standards and Sci., Daejeon, South Korea)

The fabrication of a single element transducer made from Lithium Niobate (LiNbO3) operating at 1 MHz is focused on this paper. The air-backed $LiNbO_3$ transducer is developed to be used as the standard transfer ultrasonic transducer to calibrate the ultrasound power-meter, which is measured the total emitted acoustic power radiated from the medical equipment. To clarify the precision of the acoustic power, the primary standard calibration measurement (radiation force balance, RFB) based on IEC 61161 is used to investigate the fabricated transducer. The geometry of the piezoelectric active element was first designed by the prediction of Krimholtz, Leedom, and Matthaei (KLM) simulation technique. The electrical impedance measurements of the $LiNbO_3$ element, before and after assembling into the transducer, were checked and compared. The results of electrical impedance show that the operating frequency is in the range from 1 MHz to 10 MHz by forming harmonics. The evaluations of total emitted power and radiation conductance of fabricated transducer were also revealed. Results of acoustic power have been responding up to 2.1 W, which can be assessed within 6% of expanded uncertainty ($k = 2$).

1:30

1pBA2. Sustained acoustic medicine for stimulation of wound healing: A translational research report. Matthew D. Langer and George K. Lewis (ZetrOZ, 56 Quarry Rd., Trumbull, CT 06611, mlanger@zetroz.com)

The healing of both acute and chronic wounds is a challenging clinical issue affecting more than 6.5 million Americans. The regeneration phase of wound healing is critical to restoration of function, but is often prolonged because of the adverse environment for cell growth. Therapeutic ultrasound increases nutrient absorption by cells, accelerates cellular metabolism, and stimulates production of ECM proteins, which all increase the rate of wound healing. To test the effect of long duration ultrasound exposure, an initial study of wound healing was conducted in a rat model, with wounds sutured to prevent closure via contraction. In this study, a 6 mm wound healed in 9 ± 2 days when exposed to 6 hours of ultrasound therapy, and 15 ± 1 days with a placebo device ($p<0.01$). Following IRB approval of a similar protocol for use in humans, a case study was performed on the wound closure of a chronic wound. Four weeks of daily LITUS therapy reduced the wound size by 90% from its size after 21 days of treatment with standard of care. These results demonstrate the efficacy of long duration LITUS for healing wounds in an animal model and an initial case of healing in a human subject.

1:45

1pBA3. Long duration ultrasound facilitates delivery of a therapeutic agent. Kelly Stratton, Rebecca Taggart, and George K. Lewis (ZetrOZ, 56 Quarry Rd., Trumbull, CT 06611, george@zetroz.com)

The ability for ultrasound to enhance drug delivery through the skin has been established in an animal model. This research tested the delivery of a therapeutic agent into human skin using sustained ultrasonic application over multiple hours. An IRB-approved pilot study was conducted using hyaluronan, a polymer found in the skin and associated with hydration. To assess the effectiveness of the delivery, a standard protocol was applied to measure moisture of the volar forearm with a corneometer. Fifteen subjects applied the hyalaronan to their forearms daily. One location was then treated with a multi-hour ultrasonic treatment, and the other was not. Baseline skin hydration measurements were taken for one week, followed by daily treatments with moisturizer and corneometer measurements twice per week for three weeks. Subjects experienced double the increase in sustained moisture when ultrasound was used in conjunction with a moisturizer when compared to moisturizer alone ($p<0.001$) over the four weeks. This study successfully demonstrated ultrasound treatment enhanced delivery of a therapeutic agent into the skin.

2:00

1pBA4. Characterizing the pressure field in a modified flow cytometer quartz flow cell: A combined measurement and model approach to validate the internal pressure. Camilo Perez (BioEng. and Ctr. for Industrial and Medical Ultrasound - Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, camipiri@uw.edu), Chenghui Wang (Inst. of Acoust., College of Phys. & Information Technol., Shaanxi Normal Univ., Xi'an, Shaanxi, China), Brian MacConaghy (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Juan Tu (Key Lab. of Modern Acoust., Nanjing Univ., Nanjing, Jiangsu, China), Jarred Swalwell (Oceanogr., Univ. of Washington, Seattle, WA), and Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

We incorporated an ultrasound transducer into a flow cytometer to "activate" microbubbles passing the laser interrogation zone (J. Acoust. Soc. Am. 126, 2954–2962, (2009)). This system allows high throughput recording of the volume oscillations of microbubbles, and has led to a new bubble dynamics model that incorporates shear thinning (Phys. Med. Biol. 58, 985–998 (2013)). Important parameters in the model include the ambient microbubble size, R_0, driving pressure, P_A, and the shell parameters χ and κ, the shell elasticity and viscosity, respectively. R_0 is obtained by calibrating the cytometer. Pressure calibration is difficult because the flow channel width (<200μm) is too small to insert a hydrophone. The objective of this study was to develop a calibration method for a 20-cycle, 1 MHz transient pressure field. The pressure field propagating through the channel and into water was compared to a 3-D FEM model. After validation, the model was used to simulate the driving pressure as input for the bubble dynamics model, leaving only χ and κ variables. This approach was used to determine the mechanical properties for different bubbles (albumin, lipid, and lyzozyme shells). Excellent fits were obtained in many cases, but not all, suggesting heterogeneity in microbubble shell parameters.

2:15

1pBA5. Entropy based detection of molecularly targeted nanoparticle ultrasound contrast agents in tumors. Michael Hughes (Int. Med./Cardiology, Washington Univ. School of Medicine, 1632 Ridge Bend Dr., St. Louis, MO 63108, mshatctrain@gmail.com), John McCarthy (Dept. of Mathematics, Washington Univ., St. Louis, MO), Jon Marsh, and Samuel Wickline (Int. Med./Cardiology, Washington Univ. School of Medicine, Saint Louis, MO)

In this study, we demonstrate the use of "joint entropy" of two random variables (X,Y) can be applied to markedly improve tumor conspicuity (where $X = f(t)$ =backscattered waveform and $Y = g(t)$ = a reference waveform; both differentiable functions). Previous studies have shown that a good initial choice of reference is a reflection of the original insonifying pulse taken from a stainless-steel reflector. Using this choice, joint entropy analysis is more sensitive to accumulation of targeted contrast agents than conventional gray-scale or signal energy analysis by roughly a factor of 2 [Hughes, M. S., *et al*., J. Acoust. Soc. Am., 133(1), p 283, 2013]. We now derive an improved reference that is applied to three groups of (MDA-435, breast tumor) flank tumor-implanted athymic nude mice to identify tumor vasculature after binding perfluorocarbon nanoparticles (~250 nm) to neovascular avb3 integrins. Five mice received i.v.avb3-targeted nanoparticles, five received nontargeted nanoparticles, and five received saline at a dose of 1 ml/kg, which was allowed to circulate for up to two hours prior to imaging. Three analogous groups of nonimplanted mice were imaged in the same region following the same imaging protocol. Our results indicate an improvement in contrast by a factor of 2.5 over previously published results. Thus, judicious selection of the reference waveform is critical to improving contrast-to-noise in tumor environments when attempting to detect targeted nanostructures for molecular imaging of sparse features.

2:30

1pBA6. Effects of fluid medium flow and spatial temperature variation on acoustophoretic motion of microparticles in microfluidic channels. Zhongzheng Liu and Yong-Joe Kim (Texas A&M Univ., 3123 TAMU, College Station, TX 77843, liuzz008@tamu.edu)

Current, state-of-the-art models of acoustophoretic forces, applied to microparticles suspended in fluid media inside microfluidic channels, and acoustic streaming velocities inside the microfluidic channels have been mainly derived with the assumption of "static" fluid media with uniform temperature distributions. Therefore, it has been challenging to understand the effects of "moving" fluid media and fluid medium temperature variation on acoustophoretic microparticle motion in the microfluidic channels. Here, a numerical modeling method to accurately predict the acoustophoretic motion of compressible microparticles in the microfluidic channels is presented to address the aforementioned challenge. In the proposed method, the Mass, Momentum, and Energy Conservation Equations and the State Equation are decomposed by using a perturbation method into the zeroth- to the second-order equations. Here, the fluid medium flow and temptation variation are considered in the zeroth-order equations and the solutions of the zeroth-order equations (i.e., the zeroth-order fluid medium velocities and temperature distribution) are propagated into the higher-order equations, ultimately affecting the second-order acoustophoretic forces and acoustic streaming velocities. The effects of the viscous fluid medium flow and the medium temperature variation on the acoustophoretic forces and the acoustic streaming velocities were then studied in this article by using the proposed numerical modeling method.

2:45

1pBA7. Thrombolytic efficacy and cavitation activity of rt-PA echogenic liposomes versus Definity exposed to 120-kHz ultrasound. Kenneth B. Bader, Guillaume Bouchoux, Christy K. Holland (Internal Medicine, Univ. of Cincinnti, 231 Albert Sabin Way, CVC 3933, Cincinnati, OH 45267-0586, Kenneth.Bader@uc.edu), Tao Peng, Melvin E. Klegerman, and David D. McPherson (Internal Medicine, Univ. of Texas Health Sci. Ctr., Houston, TX)

Echogenic liposomes can be used as a vector for co-encapsulation of the thrombolytic drug rt-PA and microbubbles. These agents can be acoustically activated for localized cavitation-enhanced drug delivery. The objective of our study was to characterize thrombolytic efficacy and sustained cavitation nucleation and activity from rt-PA-loaded echogenic liposomes (t-ELIP). A spectrophotometric method was used to determine the enzymatic activity of rt-PA released from t-ELIP and compared to unencapsulated rt-PA. The thrombolytic efficacy of t-ELIP, rt-PA alone, or rt-PA and the commercial contrast agent Definity® exposed to sub-megahertz ultrasound was determined in an *in vitro* flow model. Ultraharmonic (UH) emissions from stable cavitation were recorded during insonation. Both UH emissions and thrombolytic efficacy were significantly greater for rt-PA and Definity® over either rt-PA alone or t-ELIP with equivalent rt-PA loading. Furthermore, the enzymatic activity of t-ELIP was significantly lower than free rt-PA. When the dosage of t-ELIP was adjusted to compensate for the lack of enzymatic activity, similar thrombolytic efficacy was found for t-ELIP and Definity® and rt-PA. However, sustained ultraharmonic emissions were not observed for t-ELIP in the flow phantom.

3:00

1pBA8. Temporal stability evaluation of fluorescein-nanoparticles loaded on albumin-coated microbubbles. Marianne Gauthier (Dept. of Elec. and Comput. Eng., BioAcoust. Res. Lab., Univ. of Illinois at Urbana-Champaign, 4223 Beckman Inst.,405 N. Mathews, Urbana, IL 61801, frenchmg@illinois.edu), Jamie R. Kelly (Dept. of BioEng., BioAcoust. Res. Lab., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and William D. O'Brien (Dept. of Elec. and Comput. Eng., BioAcoust. Res. Lab., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Purpose: This study aims to evaluate the temporal stability of newly designed FITC-nanoparticles (NPs) loaded on albumin-coated microbubbles (MBs) to be used for future drug delivery purposes. Materials and Methods: MBs (3.6 108 MB/mL) were obtained by sonicating 5% bovine serum

albumin and 15% dextrose solution. NPs (5 mg/mL) were produced from fluorescein (FITC)-PLA polymers and functionalized using EDC/NHS. NP-loaded MBs resulted from the covalent linking between functionalized NPs and MBs via carbodiimide technique. Three parameters were quantitatively monitored over a 4-week duration at 8 time points: MB diameter was determined using a circle detection routine based on the Hough transform, MB number density was evaluated using a hemocytometer, and NP-loading yield was assessed based on the loaded-MB fluorescence uptake. Based on the hypotheses, analyses of variance or Kruskal Wallis test were run to evaluate the stability of these physical parameters over the time of the experiment. Results: Statistical analysis exhibited no significant differences in NP-loaded MB mean sizes, number densities, and loading yields over time ($p > 0.05$). Conclusion: Newly designed NP-loaded MBs are stable over at least a 4-week duration and can be used without extra precaution concerning their temporal stability. [This work was supported by NIH R37EB002641.]

3:15–3:30 Break

3:30

1pBA9. Chronotropic effect in rats heart caused by pulsed ultrasound. Olivia C. Coiado and William D. O'Brien Jr. (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, 4223 Beckman Inst., Urbana, IL 61801, oliviacoiado@hotmail.com)

This study investigated the dependence of an increasing/decreasing sequence of pulse repetition frequencies (PRFs) on the chronotropic effect via the application of 3.5-MHz pulsed ultrasound (US) on the rat heart. The experiments were divided into three 3-month-old female rat groups ($n = 4$ ea): control, PRF increase and PRF decrease. Rats were exposed to transthoracic ultrasonic pulses at ~0.50% of duty factor at 2.0-MPa peak rarefactional pressure amplitude. For the PRF increase group, the PRF started lower than that of the rat's heart rate and was increased sequentially in 1-Hz steps every 5 s (i.e., 4, 5, and 6 Hz) for a total duration of 15 s. For the PRF decrease group, the PRF started greater than that of the rat's heart rate and was decreased sequentially in 1-Hz steps every 5 s (i.e., 6, 5, and 4 Hz). For the PRF decrease and control groups, the ultrasound application resulted in a significant negative chronotropic effect (~11%) after ultrasound exposure. However, for the PRF increase group, a significant but less decrease of the heart rate (~3%) was observed after ultrasound exposure. The ultrasound application caused a negative chronotropic effect after US exposure for increase/decrease US group. [Support: NIH Grant R37EB002641.]

3:45

1pBA10. Ultrasonic welding in orthopedic implants. Kristi R. Korkowski and Timothy Bigelow (Mech. Eng., Iowa State Univ., 2201 Coover Hall, Ames, IA 50011, korkowsk@iastate.edu)

A critical event in hip replacement is the occurrence of osteolysis. Cemented hip replacements most commonly use polymethylmethacrylate (PMMA), not as an adhesive but rather a filler to limit micromotion and provide stability. PMMA, however, contributes to osteolysis through both a thermal response during curing and implant wear debris. In order to mitigate the occurrence of osteolysis, we are exploring ultrasonic welding as a means of attachment. Weld strength was assessed using ex vivo bovine rib and femur bones. A flat end mill provided 20 site locations for insertion of an acrylonitrile butadiene styrene, ABS pin. Each location was characterized on topography, porosity, discoloration, and any other notable features. Each site was welded using a Branson Ultrasonic Welder 2000iw; 20 kHz, 1100 W. Machine parameters include weld force, weld time, and hold time. The bond strength was determined using a tensile tester. Tensile testing showed a negative correlation between porosity and bond strength. Further evaluation and characterization of bone properties to bond strength will enable appropriate selection of welding properties to ensure a superior bond.

4:00

1pBA11. Estimation of subsurface temperature profiles from infrared measurements during ultrasound ablation. Tyler R. Fosnight, Fong Ming Hooi, Sadie B. Colbert, Ryan D. Keil, and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Measurement of *in situ* spatiotemporal temperature profiles would be useful for developing and validating thermal ablation methods and therapy monitoring approaches. Here, finite difference and analytic solutions to Pennes' bio-heat transfer equation were used to determine spatial correlations between temperature profiles on parallel planes. Time delays and scale factors for correlated profiles were applied to infrared surface-temperature measurements to estimate subsurface temperatures. To test this method, *ex vivo* bovine liver tissue was sonicated by linear image-ablate arrays with 1–6 pulses of 5.0 MHz unfocused (7.5 s, 64.4–92.0 W/cm^2 *in situ* I_{SPTP}) or focused (1 s, 562.7–799.6 W/cm^2 *in situ* I_{SPTP}, focus depth 10 mm) ultrasound. Temperature was measured on the liver surface by an infrared camera at 1 fps and extrapolated to the imaging/ablation plane, 3 mm below the surface. Echo decorrelation maps were computed from pulse-echo signals captured at 118 fps during 5.0 s rest periods beginning 1.1 s after each sonication pulse. Tissue samples were frozen at -80 °C, sectioned, vitally stained, imaged, and segmented for analysis. Estimated thermal dose profiles showed correspondence with segmented tissue histology, while thresholded temperature profiles corresponded with measured echo decorrelation. These results suggest utility of this method for thermal ablation research.

4:15

1pBA12. Temperature dependence of harmonics generated by nonlinear ultrasound beam propagation in water. Borna Maraghechi, Michael C. Kolios, and Jahan Tavakkoli (Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, borna.maraghechi@ryerson.ca)

Ultrasound thermal therapy is used for noninvasive treatment of cancer. For accurate ultrasound based temperature monitoring in thermal therapy, the temperature dependence of acoustic parameters is required. In this study, the temperature dependence of acoustic harmonics was investigated in water. The pressure amplitudes of the transmitted fundamental frequency (p1), and its harmonics (second (p2), third (p3), fourth (p4), and fifth (p5)) generated by nonlinear ultrasound propagation were measured by a calibrated hydrophone in water. The hydrophone was placed at the focal point of a focused 5-MHz transducer (f-number 4.5) to measure the acoustic pressure. Higher harmonics were generated by transmitting a 5-MHz 15-cycle pulse that resulted in a focal positive peak pressure of approximately 0.26 MPa in water. The water temperature was increased from 26 °C to 52 °C in increments of 2 °C. Due to this temperature elevation, the value of p1 decreased by 9%±1.5% (compared to its value at 26 °C) and values of p2, p3, p4, and p5 increased by %5±2%, 22%±8%, 44%±7%, and 55%±5%, respectively. The results indicate that the nonlinear harmonics are highly temperature dependent and their temperature sensitivity increase with the harmonic number. It is concluded that the nonlinear harmonics could potentially be used for ultrasound-based thermometry.

4:30

1pBA13. Implementation of a perfectly matched layer in nonlinear continuous wave ultrasound simulations. Xiaofeng Zhao and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, zhaoxia6@msu.edu)

FOCUS, the "Fast Object-Oriented C++ Ultrasound Simulator" (http://www.egr.msu.edu/~fultras-web), simulates nonlinear ultrasound propagation by numerically evaluating the Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation. For continuous-wave excitations, KZK simulations in FOCUS previously required that the simulations extend over large radial distances relative to the aperture radius, which reduced the effect of reflections from the boundary on the main beam. To reduce the size of the grid required for these calculations, a perfectly matched layer (PML) was recently added to the KZK simulation routines in FOCUS. Simulations of the linear pressure fields generated by a spherically focused transducer with an aperture radius of 1.5 cm and a radius of curvature of 6cm are evaluated for a peak surface pressure of 0.5 MPa and a 1 MHz fundamental frequency.

Results of linear KZK simulations with and without the PML are compared to an analytical solution of the linear KZK equation on-axis, and the results show that simulations without the PML require a radial boundary that is at least seven times the aperture radius, whereas the PML enables accurate simulations for a radial boundary that is only two times the aperture radius. [This work was supported in part by NIH Grant R01 EB012079.]

4:45

1pBA14. An improved time-base transformation scheme for computing waveform deformation during nonlinear propagation of ultrasound. Boris de Graaff, Shreyas B. Raghunathan, and Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628CJ, Netherlands, m.d.verweij@tudelft.nl)

Nonlinear propagation plays an important role in various applications of medical ultrasound, like higher harmonic imaging and high intensity focused ultrasound (HIFU) treatment. Simulation of nonlinear ultrasound fields can greatly assist in explaining experimental observations and in predicting the performance of novel procedures and devices. Many numerical simulations are based on the generic split-step approach, which takes the ultrasound field at the transducer plane and propagates this forward over successive parallel planes. Usually, the spatial steps between the planes are small and the diffraction, attenuation, and nonlinear deformation may be treated as separate substeps. For the majority of methods, e.g., for all KZK-type methods, the nonlinear substep relies on the implicit solution of the one-dimensional Burgers equation, which is implemented using a time-base transformation. This generally works fine, but when the shock wave regime is approached, reduced spatial steps are required to avoid time points to "cross over," and the method can become notoriously slow. This paper analyses the fundamental difficulty with the common time base transformation, and provides an alternative that does not suffer from the mentioned slowdown. Numerical results will be shown to demonstrate that this alternative will allow much larger spatial steps without compromising the numerical accuracy.

5:00

1pBA15. An error reduction algorithm for numeric calculation of the spatial impulse response. Nils Sponheim (Inst. of Industrial Dev., Faculty of Technol., Art and Design, Oslo and Akershus Univ. College of Appl. Sci., Pilestredet 35, P.O. Box 4, St. Olavs plass, Oslo NO-0130, Norway, nils.sponheim@hioa.no)

The most frequently used method for calculation of the pulsed pressure field of ultrasonic transducers is the spatial impulse response (SIR) method. This paper presents a new numeric approach that reduce the numeric error by weighting the contribution of each source element into the SIR time array, by considering the exact time of arrival of each contribution. The resolution of the time array Δt must be finite. This results in an error in travel time of $\pm\Delta t/2$. However, we know the exact travel time and based on this, we can share the contribution from each source element between the two closest time elements so that the average time corresponds to the exact travel time and thereby reduce the numeric error. This study compares the old and the new numeric algorithm with the analytic solution for a planar circular disk because it has a simple analytic solution. The paper presents calculations of the SIR for selected points in space and calculations of the RMS-error between the numeric algorithms and the analytic solution. The proposed new numeric algorithm decreases the numeric noise or error with a factor of 5 compared to the old numeric algorithm.

5:15

1pBA16. Teaching auscultation visually with low cost system, is it feasible? Sergio L. Aguirre (Universidade Federal de Santa Maria, Rua Professor Heitor da Graça Fernandes, Avenida Roraima 1000 Centro de Tecnologia, Santa Maria, Rio Grande do Sul 97105-170, Brazil, sergio.aguirre@eac.ufsm.br), Ricardo Brum, Stephan Paul, Bernardo H. Murta, and Paula P. Jardin (Universidade Federal de Santa Maria, Santa Maria, RS, Brazil)

Cardiac auscultation can generate important information in the diagnosis of diseases. The sounds that the cardiac system provides are understood in the frequency range of human hearing, but in a region of low sensitivity. This project aims to build a low cost didactic software/hardware set for teaching cardiac auscultation technique in Brazilian universities. The frequencies of interest to describe the human cardiac cycle were found in the range of 20 Hz to 1 kHz which includes low frequencies where available low-cost transducers usually have large errors. To create the system, an optimization of the geometry of the chestpiece is being programmed with finite element simulations; meanwhile, digital filters for specific frequencies of interest and an interface based on MATLAB are being developed. There were needed filters for the gallops (20 to 70 Hz), heart beats (20 to 100 Hz), ejection murmurs (100 to 500 Hz), mitral stenosis (30 to 80 Hz), and regurgitations (200 to 900 Hz). The FEM simulation of a chestpiece demonstrates high signaling levels on the desired frequency range, which can be used with the filters to obtain specific information. Furthermore, the ideal signal recording equipments will be defined, implemented, and tested.

Session 1pNS

Noise and Physical Acoustics: Metamaterials for Noise Control II

Olga Umnova, Cochair
University of Salford, The Crescent, Salford M5 4WT, United Kingdom

Keith Attenborough, Cochair
DDEM, The Open University, Walton Hall, Milton Keynes MK7 6AA, United Kingdom

Chair's Introduction—12:55

Invited Paper

1:00

1pNS1. Sound propagation in the presence of a resonant surface. Logan Schwan (Univ. of Salford, The Crescent, Salford m5 4wt, United Kingdom, logan.schwan@gmail.com) and Olga Umnova (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom)

The interactions between acoustic waves and an array of resonators are studied. The resonators are arranged periodically on an impedance surface so that the scale separation between sound wavelength and the array period is achieved. An asymptotic multi-scale model which accounts for viscous and thermal losses in the resonators is developed and is used to derive an effective surface admittance. It is shown that the boundary conditions at the surface are substantially modified around the resonance frequency. The pressure field on the surface is nearly canceled leading to a phase shift between the reflected and the incident waves. The array can also behave as an absorbing layer. The predictions of the homogenized model are compared with multiple scattering theory (MST) applied to a finite size array and the limitations of the former are identified. The influence of the surface roughness and local scattering on the reflected wave is discussed.

Contributed Papers

1:20

1pNS2. Flexural wave induced coherent scattering in arrays of cylindrical shells in water. Alexey S. Titovich and Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, alexey17@eden.rutgers.edu)

A periodic array of elastics shells in water is a sonic crystal with local resonances in the form of flexural vibrations. This acoustic metamaterial has seen application in wave steering by grading the index in the array, as well as acoustic filters manifested by Bragg scattering. The primary reason for using shells is that they can be tuned quasi-statically to have water-like effective acoustic properties. The issue is that the modally dense flexural resonances can form pseudogaps in the frequency response resulting in total reflection from the array. Furthermore, if a flexural resonance falls in the Bragg band gap, total transmission is possible at that frequency. Although the scattered wave due to low order flexural vibration of a thin shell is evanescent, when several shells are closely spaced, the effect on the far-field response is dramatic. In this paper, the interaction of neighboring shells is investigated theoretically using the Love-Timoshenko shell theory and multiple scattering. A simple model is offered to describe the interaction of modes based on the analytical work. The directionality of the lowest flexural modes is also discussed as it can lead to phasing between neighboring shells.

1:35

1pNS3. A thin-panel underwater acoustic absorber. Ashley J. Hicks, Michael R. Haberman, and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs, Univ. of Texas at Austin, 3607 Greystone Dr., Apartment 1410, Austin, TX 78731, a.jean.hicks@utexas.edu)

We present experimental results on the acoustic behavior of thin-panel underwater sound absorbers composed of a sub-wavelength layered structure. The panels are formed using an inner layer of Delrin or PLA plastic with circular air-filled holes sandwiched between two rubber outer layers. The panel structure mimics a planar encapsulated bubble screen exactly one bubble thick, but displays performance that is significantly more broadband than a comparable bubble screen, which is only useful near the resonance frequency of the bubble. Initial results indicate 10 dB of insertion loss in the frequency range 1 kHz to 5 kHz for a panel that is about 1/250th of a wavelength in thickness at the lowest frequency. The effect of air volume fraction and the use of a 3-D printed (porous) inner layer on insertion loss will be presented and discussed. [Work supported by ONR.]

1:50

1pNS4. Micromechanical effective medium modeling of metamaterials of the Willis form. Michael B. Muhlestein, Michael R. Haberman, and Preston S. Wilson (Appl. Res. Labs. and Dept. of Mech. Eng., Univ. of Texas at Austin, 3201 Duval Rd. #928, Austin, TX 78759, mimuhle@gmail.com)

The unique behavior of acoustic metamaterials (AMM) results from deeply sub-wavelength structures with hidden degrees of freedom rather than the inherent material properties of their constituents. This distinguishes AMM from classical composite or cellular materials and also complicates attempts to model their overall response. This is especially true when sub-wavelength structures yield anisotropic effective material response, a key feature of AMM devices designed using transformation acoustics. Further, previous work has shown that the dynamic response of heterogeneous materials must include coupling between the overall strain and momentum fields [Milton and Willis, Proc. R. Soc. A **463**, 855–880, (2007)]. A micromechanical homogenization model of the overall Willis constitutive equations is presented to address these difficulties. The model yields a low-volume-fraction estimate of anisotropic and frequency-dependent effective properties in

the long-wavelength limit. The model employs volume averages of the dyadic Green's function calculating the particle displacement resulting from a unit force source. This Green's function is shown to be analogous to one that determines the particle velocity in a fluid resulting from a unit dipole moment. The predicted effective properties for isotropic materials with spherical inclusions fall within the Hashin-Shtrikman bounds and agree with self-consistent estimates. [Work supported by ONR.]

2:05

1pNS5. Acoustic metamaterial homogenization based on equivalent fluid media with coupled field response. Caleb F. Sieck (Appl. Res. Labs. and Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, 4021 Steck Ave #115, Austin, TX 78759, cfsieck@utexas.edu), Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Andrea Alù (Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

Homogenization schemes for wave propagation in heterogeneous electromagnetic (EM) and elastic materials indicate that EM bianisotropy and elastic momentum-strain and stress-velocity field coupling is required to correctly describe the effective behavior of the medium [Alu, Phys. Rev. B, **84**, 075153 (2011); Milton and Willis, Proc. R. Soc. A, **463**, 855–880, (2007)]. Further, the determination of material coupling terms in EM resolves apparent violations of causality and passivity which is present in earlier models [A. Alù, Phys. Rev. B, **83**, 081102(R) (2011)]. These details have not received much attention in fluid acoustics, but they are important for a proper description of acoustic metamaterial behavior. We derive expressions for effective properties of a heterogeneous fluid medium from expressions for the conservation of mass, the conservation of momentum, and the equation of state and find a physically meaningful effective material response from first-principles. The results show inherent coupling between the ensemble averaged volume strain-momentum and pressure-velocity field. The approach is valid for an infinite periodic lattice of heterogeneities and employs zero-, first-, and second-order tensorial Green's functions to relate point-discontinuities in compressibility and density to far field pressure and particle velocity fields. [This work was supported by the Office of Naval Research.]

2:20

1pNS6. Nonlinear behavior of a coupled multiscale material containing snapping acoustic metamaterial inclusions. Stephanie G. Konarski, Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu)

Snapping acoustic metamaterial (SAMM) inclusions are engineered subwavelength structures that exhibit regimes of both positive and negative stiffness. Snapping is defined as large, rapid deformations resulting from the application of an infinitesimal change in externally applied pressure. This snapping leads to a large hysteretic response at the inclusion scale and is thus of interest for enhancing absorption of energy in acoustic waves. The research presented here models the forced dynamics of a multiscale material consisting of SAMM inclusions embedded in a nearly incompressible viscoelastic matrix material to explore the influence of small-scale snapping on enhanced macroscopic absorption. The microscale is characterized by a single SAMM inclusion, while the macroscale is sufficiently large to encompass a low volume fraction of non-interacting SAMM inclusions within the nearly incompressible matrix. A model of the forced dynamical response of this heterogeneous material is achieved by coupling the two scales in time and space using a generalized Rayleigh-Plesset analysis, which has been adapted from the field of bubble dynamics. A loss factor for the heterogeneous medium is examined to characterize energy dissipation due to the forced behavior of these metamaterial inclusions. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and Office of Naval Research.]

2:35

1pNS7. Cloaking of an acoustic sensor using scattering cancelation. Matthew D. Guild (Dept. of Electronics Eng., Universitat Politecnica de Valencia, Camino de vera s/n (Edificio 7F), Valencia 46022, Spain, mdguild@utexas.edu), Andrea Alù (Dept. of Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Acoustic scattering cancelation (SC) is an approach enabling the elimination of the scattered field from an object, thereby cloaking it, without restricting the incident wave from interacting with the object. This aspect of an SC cloak lends itself well to applications in which one wishes to extract energy from the incident field with minimal scattering, such as for sensing and noise control. In this work, an acoustic cloak designed based on the scattering cancelation method, and made of two effective fluid layers, is applied to the case of an acoustic sensor consisting of a hollow piezoelectric shell with mechanical absorption, providing a 20–50 dB reduction in the scattering strength. The cloak is shown to increase the range of frequencies over which there is nearly perfect phase fidelity between the acoustic signal and the voltage generated by the sensor, while remaining within the physical bounds of a passive absorber. The feasibility of achieving the necessary fluid layer properties is demonstrated using sonic crystals with the use of readily available acoustic materials. [Work supported by the US ONR and Spanish MINECO.]

2:50

1pNS8. Cloaking non-spherical objects and collections of objects using the scattering cancelation method. Ashley J. Hicks (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Appl. Res. Labs., 10000 Burnet Rd., Austin, TX 78758, ahicks@arlut.utexas.edu), Matthew D. Guild (Wave Phenomena Group, Dept. of Electronics Eng., Universitat Politècnica de València, Valencia, Spain), Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Andrea Alù (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Acoustic cloaks can be designed using transformation acoustics (TA) to guide acoustic disturbances around an object. TA cloaks, however, require the use of exotic materials such as pentamode materials [Proc. R. Soc. A. **464**, pp. 2411–2434, (2008)]. Alternatively, the scattering cancelation (SC) method allows the cloaked object to interact with the acoustic wave and can be realized with isotropic materials [Phys. Rev. B., **86**, 104302 (2012)]. Unfortunately, SC cloaking performance may be degraded if the shape of the cloaked object diverges from the one for which the cloak was originally designed. This study investigates the design of two-layer SC cloaks for imperfect spherical objects. The cloaking material properties are determined by minimizing the scattered field from a model of the imperfect object approximated as a series of concentric shells. Predictions from this approximate analytical model are compared with three-dimensional finite element (FE) models of the cloaked and uncloaked non-spherical shapes. Analytical and FE results are in good agreement for $ka \leq 5$, indicating that the SC method is robust to object imperfections. Finally, FE models are used to explore SC cloak robustness to multiple-scattering by investigating linear arrays of cloaked objects for different incident angles. [Work supported by ONR.]

3:05

1pNS9. Parity-time symmetric metamaterials and metasurfaces for loss-immune and broadband acoustic wave manipulation. Romain Fleury, Dimitrios Sounas, and Andrea Alu (ECE Dept., The Univ. of Texas at Austin, 1 University Station C0803, Austin, TX 78712, romain.fleury@utexas.edu)

We explore the largely uncharted scattering properties of acoustic systems that are engineered to be invariant under a special kind of space-time

symmetry, consisting in taking their mirror image and running time backwards. Known as Parity-Time symmetry, this special condition is shown here to lead to acoustic metamaterials that possess a balanced distribution of gain (amplifying) and loss (absorbing) media, at the basis of ideal loss-compensation, and under certain conditions, unidirectional invisibility. We have designed and built the first acoustic metamaterial with parity-time symmetric properties, obtained by pairing the acoustic equivalent of a lasing system with a coherent perfect acoustic absorber, implemented using electro-acoustic resonators loaded with non-Foster electrical circuits. The active system can be engineered to be fully stable and, in principle, broadband. We discuss the underlying physics and present the realization of a unidirectional invisible acoustic sensor with unique sensing properties. We also discuss the potential of PT acoustic metamaterials and metasurfaces for a variety of metamaterial-related applications, which we obtain in a loss-immune and broadband fashion, including perfect cloaking of sensors, planar focusing, and unidirectional cloaking of large objects.

MONDAY AFTERNOON, 27 OCTOBER 2014 — INDIANA C/D, 1:15 P.M. TO 4:45 P.M.

Session 1pPA

Physical Acoustics and Noise: Jet Noise Measurements and Analyses II

Richard L. McKinley, Cochair
Battlespace Acoustics, Air Force Research Laboratory, 2610 Seventh Street, Wright-Patterson AFB, OH 45433-7901

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

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Chair's Introduction—1:15

Invited Papers

1:20

1pPA1. Considerations for array design and inverse methods for source modeling of full-scale jets. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Blaine M. Harker, Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC) and Richard L. McKinley (Air Force Res. Lab., Boston, OH)

Microphone array-based measurements of full-scale jet noise sources necessitate the adaptation and incorporation of advanced array processing methodologies. Arrays for full-scale jets measurements can require large apertures, high spatial sampling densities, and strategies to account for partially coherent fields. Many approaches have been taken to sufficiently capture radiated noise in past jet noise investigations, including patch-and-scan measurements with a small dense array, one-dimensional measurements along the extent of the jet in conjunction with an axisymmetric assumption, and full two-dimensional source coverage with a large microphone set. Various measurement types are discussed in context of physical jet noise field properties, such as spatial coherence, source stationary, and frequency content.

1:40

1pPA2. Toward the development of a noise and performance tool for supersonic jet nozzles: Experimental and computational results. Christopher J. Ruscher (Spectral Energies, LLC, 2654 Solitaire Ln. Apt. #3, Beavercreek, OH 45431, cjrusche@gmail.com), Barry V. Kiel (RQTE, Air Force Res. Lab., Dayton, OH), Sivaram Gogineni (Spectral Energies, LLC, Dayton, OH), Andrew S. Magstadt, Matthew G. Berry, and Mark N. Glauser (Dept. of Mech. and Aerosp. Eng., Syracuse Univ., Syracuse, NY)

Modal decomposition of experimental and computational data for a range of two- and three-stream supersonic jet nozzles will be conducted to study the links between the near-field flow features and the far-field acoustics. This is accomplished by decomposing near-field velocity and pressure data using proper orthogonal decomposition (POD). The resultant POD modes are then used with the far-field sound to determine a relationship between the near-field modes and portions of the far-field spectra. A model will then be constructed for each of the fundamental modes, which can then be used to predict the entire far-field spectrum for any supersonic jet. The resultant jet noise model will then be combined with an existing engine performance code to allow parametric studies to optimize thrust, fuel consumption, and noise reduction.

2:00

1pPA3. Finely resolved spatial variation in F-22 spectra. Tracianne B. Neilsen, Kent L. Gee, Hsin-Ping C. Pope, Blaine Harker (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Examination of the spatial variation in the spectrum from ground-based microphones near an F-22 Raptor has revealed spectral features at high engine power that are not seen at intermediate power or in laboratory-scale jet noise. At military and afterburner powers, a double peaked spectrum is detected around the direction of maximum radiation. In this region, there is not a continuous variation in peak frequency with downstream distance, as seen in lab-scale studies, but a transition between the relative levels for two discrete one-third octave bands. Previous attempts to match similarity spectra for turbulent mixing noise to a few of these measurements split the difference between the two peak frequencies [Neilsen *et al.*, J. Acoust. Soc. Am. 133, 2116–2125 (2013)]. The denser spatial resolution afforded by examining the spectral variation on all 50 ground-based microphones, located 11.6 m to the sideline and spanning 30 m, provides the opportunity to further investigate this phenomenon and propose a more complete formulation of expected spectral shapes. Special care must be given to account for the relative amount of waveform steepening, which varies with level, distance, and angular position. [Work supported by ONR.]

2:20

1pPA4. Experimental and computational studies of noise reduction for tactical fighter aircraft. Philip Morris, Dennis K. McLaughlin, Russell Powers, Nidhi Sikarwar, and Matthew Kapusta (Aerosp. Eng., Penn State Univ., 233C Hammond Bldg., University Park, PA 16802, pjm@psu.edu)

The noise levels generated by tactical fighter aircraft can result in Noise Induced Hearing Loss for Navy personnel, particularly those involved in carrier deck operations. Reductions in noise source levels are clearly necessary, but these must be achieved without a loss in aircraft performance. This paper describes an innovative noise reduction technique that has been shown in laboratory scale measurements to provide significant reductions in both mixing as well as broadband shock-associated noise. The device uses the injection of relatively low pressure and low mass flow rate air into the diverging section of the military-style nozzle. This injection generates “fluidic inserts” that change the effective nozzle area ratio and generate streamwise vorticity that breaks up the large scale turbulent structures in the jet exhaust that are responsible for the dominant mixing noise. The paper describes noise measurements with and without forward flight that demonstrate the noise reduction effectiveness of the inserts. The experiments are supported by computations that help to understand the flow field generated by the inserts as well as help to optimize the distribution and strength of the flow injection.

2:40

1pPA5. Detection and analysis of shock-like waves emitted by heated supersonic jets using shadowgraph flow visualization. Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of MS, 1 Coliseum Dr., University, MS 38677, nmurray@olemiss.edu)

Shock-like waves in the acoustic field adjacent to the shear layer formed by a supersonic, heated jet are observed using the method of retro-reflective shadowgraphy. The two inch diameter jet issued from a converging–diverging nozzle at a pressure ratio of 3.92 with a temperature ratio of 3.3. Image sets were obtained near the jet exit and in the post-potential core region. In both locations, shock-like waves can be observed immediately adjacent to the jet shear layer. Each image is subdivided into a set of overlapping tiles. A radon transform is applied to the auto-correlation of each tile providing a quantitative measure of the dominant propagation direction of waves in each sub-region. The statistical distribution of propagation angles over the image space provides a measure of the distribution of source convection speeds and source locations in the jet shear layer. Results show general agreement with a convection speed on the order of 70 percent of the jet velocity.

3:00–3:20 Break

3:20

1pPA6. Where are the nonlinearities in jet noise? Charles E. Tinney (Ctr. for AeroMech. Res., The Univ. of Texas at Austin, ASE/EM, 210 East 24th St., Austin, TX 78712, cetinney@utexas.edu) and Woutijn J. Baars (Mech. Eng., The Univ. of Melbourne, Parkville, VIC, Australia)

For some time now it has been theorized that spatially evolving instability waves in the irrotational near-field of jet flows couple both linearly and nonlinearly to generate far-field sound [Sandham and Salgado, Philos. Trans. R. Soc. Am. 366 (2008); Suponitsky, J. Fluid Mech. 658 (2010)]. An exhaustive effort at The University of Texas of Austin was initiated in 2008 to better understand this phenomenon, which included the development of a unique analysis technique for quantifying their coherence [Baars *et al.*, AIAA Paper 2010–1292 (2010); Baars and Tinney, Phys. Fluids 26, 055112 (2014)]. Simulated data have shown this technique to be effective, albeit, insurmountable failures arise when exercised on real laboratory measurements. The question that we seek to address is how might jet flows manifest nonlinearities? Both subsonic and supersonic jet flows are considered with simulated and measured data sets encompassing near-field and far-field pressure signals. The focus then turns to considering nonlinearities in the form of cumulative distortions, and the conditions required for them to be realized in a laboratory scale facility [Baars, *et al.*, J. Fluid Mech. 749 (2014)].

3:40

1pPA7. Characterization of supersonic jet noise and its control. Ephraim Gutmark, Dan Cuppoletti, Pablo Mora, Nicholas Heeb, and Bhupatindra Malla (Aerosp. Eng. and Eng. Mech., Univ. of Cincinnati, 799 Rhodes Hall, Cincinnati, OH 45221, gutmarej@ucmail.uc.edu)

As supersonic aircraft and their turbojet engines become more powerful they emit more noise. The principal physical difference between the jets emanating from supersonic jets and those from subsonic jets is the presence of shocks in the supersonic one. This paper summarizes a study of noise reduction technologies applied to supersonic jets. The measurements are performed with a simulated

exhaust of a supersonic nozzle representative of supersonic aircraft. The nozzle has a design Mach number of 1.56 and is examined at design and off-design conditions. Several components of noise are present including mixing noise, screech, broadband shock associated noise, and crackle. Chevrons and fluidic injection by microjets and a combination of them are shown to reduce the noise generated by the main jet. These techniques provide significant reduction in jet noise. PIV provides detailed information of the flow and brings out the physics of the noise production and reduction process.

Contributed Papers

4:00

1pPA8. Influence of windscreen on impulsive noise measurement. per rasmussen (G.R.A.S. Sound & Vib. A/S, Skovlytoften 33, Holte 2840, Denmark, pr@gras.dk)

The nearfield noise from jet engines may contain impulsive sound signals with high crest factors. Most jet engine noise measurements are performed outside in potentially windy conditions, and it may, therefore, be necessary to use windscreens on microphones to reduce the influence of wind induced noise on the microphone. The windscreen will, however, influence the frequency response of the microphone especially at high frequencies. This will change both the magnitude and the phase response and, therefore, change the measured impulse. The effect of different sizes of windscreen is investigated and the effect on impulsive type signals is evaluated both in the time domain and the frequency domain.

4:15

1pPA9. Comparison of nonlinear, geometric, and absorptive effects in high-amplitude jet noise propagation. Brent O. Reichman, Kent L. Gee, Tracianne B. Neilsen, Joseph J. Thaden (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), and Michael M. James (Blue Ridge Research and Consulting, LLC, Asheville, NC)

In recent years, understanding of nonlinearity in noise from high-performance jet aircraft has increased, with successful modeling of nonlinear propagation in the far field. However, the importance and characteristics of nonlinearity in the near field are still debated. An ensemble-averaged, frequency-domain version of the Burgers equation can be inspected to directly compare the effects of nonlinearity on the sound pressure level with the effects of atmospheric absorption and geometric spreading on a decibel scale. This nonlinear effect is calculated using the quadspectrum of the pressure and the squared pressure waveforms. Results from applying this analysis to F-22A data at various positions in the near field reveal that in the near field the nonlinear effects are of the same order of magnitude as geometric spreading and that both of these effects are significantly greater than absorption in the area of maximum radiation. [Work supported by ONR and an ORISE fellowship through AFRL.]

4:30

1pPA10. Correlation lengths in deconvolved cross-beamforming measurements of military jet noise. Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC,, Provo, UT 84602, blaineharker@byu.net), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson Air Force Base, OH), and Michael M. James (Blue Ridge Research and Consulting, LLC, Asheville, NC)

Beamforming algorithms have been applied in multiple contexts in aeroacoustic applications, but difficulty arises when applying these to the partially correlated and distributed sources found in jet noise. To measure and more accurately distinguish correlated sources, cross-beamforming methods are employed to incorporate correlation information. Deconvolution methods such as DAMAS-C, an extension of the deconvolution approach for the mapping of acoustic sources (DAMAS), remove array effects from cross-beamforming applications and further resolve beamforming results. While DAMAS-C results provide insight to correlation between sources, the extent to which these results relate to source correlation remains to be analyzed. Numerical simulations of sources with varying degrees of correlation are provided to benchmark the DAMAS-C results. Finally, correlation lengths are established for DAMAS-C results from measurements for full-scale military jet noise sources. [Work supported by ONR.]

Session 1pSCa

Speech Communication and Biomedical Acoustics: Findings and Methods in Ultrasound Speech Articulation Tracking

Keith Johnson, Cochair
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Susan Lin, Cochair
UC Berkeley, 1203 Dwinelle Hall, UC Berkeley, Berkeley, CA 94720

Chair's Introduction—1:00

Invited Papers

1:05

1pSCa1. Examining suprasegmental and morphological effects on constriction degree with ultrasound imaging. Lisa Davidson (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, lisa.davidson@nyu.edu)

Two case studies of ultrasound imaging use tongue shape differences to investigate whether suprasegmental influences affect the articulatory implementation of otherwise equivalent phonemic sequences. First, we examine whether word-medial and word-final stop codas have the same degree of constriction (e.g., "blacktop" vs. "black top"). Previous research on syllable position effects on articulatory implementation have conflated syllable position with word position, and this study investigates whether each prosodic factor has an independent contribution. Results indicate that where consistent differences are found, they are due not to the prosodic position but to speaker-specific implementation. Second, we examine whether morphological status influences the darkness of American English /l/ in comparing words like "tallest" and "flawless." While the intervocalic /l/s in "tall-est" and "flaw-less" are putatively assigned the same syllabic status, the /l/ in "tallest" corresponds to the coda /l/ of the stem "tall" whereas that of "flawless" is the onset of the affix "-less." Results indicate that /l/ is darker—the tongue is lower and more retracted—when corresponding to the coda of the stem word. Data in both studies were analyzed with smoothing spline ANOVA, an effective statistical technique for examining differences between whole tongue curves.

1:25

1pSCa2. Imaging dynamic lingual movements that we could previously only imagine. Amanda L. Miller (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, amiller@ling.osu.edu)

Pioneering lingual ultrasound studies of speech demonstrated that almost the entire tongue could be imaged (McKay 1957). Early studies contributed to our knowledge of tongue shape and tongue bracing in vowels (Morrish *et al.* 1984; Stone *et al.* 1987). However, until recently, lingual ultrasound studies have been limited to standard video frame rates of 30 fps, which are sufficient only for imaging stable speech sounds such as vowels and liquids. High frame rate lingual ultrasound (>100 fps) allows us to view the production of dynamic speech sounds, such as stop consonants, and even click consonants. The high sampling rate, which yields an image of the tongue every 8–9 ms, improves image quality, by decreasing temporal smear, allowing even tongue tip movements to be visualized to a greater extent than was previously possible. Results from several high frame rate ultrasound studies (114 fps) of consonants that were collected and analyzed using the CHAUSA method (Miller and Finch 2011) are presented. The studies elucidate (a) tongue dorsum and root gestures in velar and uvular pulmonic consonants; (b) tongue coronal, dorsal, and root gestures in four contrastive click consonants; and (c) lingual gestures in pulmonic fricatives.

1:45

1pSCa3. Ultrasound evidence for place of articulation of the mora nasal /ɴ/ in Japanese. Ai Mizoguchi (The Graduate Ctr., City Univ. of New York, 365 Fifth Ave., Rm. 7304, New York, NY 10016, amizoguchi@gc.cuny.edu) and Douglas H. Whalen (Haskins Labs., New Haven, CT)

The Japanese mora nasal /ɴ/, which occurs in syllable-final position, takes its place of articulation from the following segment if there is one. However, the mora nasal in utterance-final position is often transcribed as velar, uvular, or even placeless. The present study examines the tongue shapes in Japanese using ultrasound imaging to investigate whether Japanese mora nasal /ɴ/ is placeless and to assess whether assimilation to following segments is gradient or categorical. Preliminary results from ultrasound imaging from one native speaker of Tokyo dialect showed three shapes for final /ɴ/, even though the researchers could not distinguish them perceptually. Results from assimilation contexts showed that the velar gesture for /ɴ/ was not deleted. All gestures remained and assimilation was not categorical, even though perceptually, it was. The velar gesture for /ɴ/ might be expected to be deleted before an alveolar /n/ because they are both lingual, but a blending of the two tongue gestures occurred instead. Variability in place of articulation in final position occurred even within one speaker. Categorical assimilation was not observed in any phonological environments studied. The mora nasal may vary across speakers, so further research is needed to determine whether it behaves similarly for more speakers.

2:05

1pSCa4. A multi-modal imaging system for simultaneous measurement of speech articulator kinematics for bedside applications in clinical settings. David F. Conant (Neurological Surgery, UCSF, 675 Nelson Rising Ln., Rm. 635, San Francisco, CA 94143, dfconant@gmail.com), Kristofer E. Bouchard (LBNL, San Francisco, CA), Anumanchipalli K. Gopala, Ben Dichter, and Edward F. Chang (Neurological Surgery, UCSF, San Francisco, CA)

A critical step toward a neurological understanding of speech generation is to relate neural activity to the movement of articulators. Here, we describe a noninvasive system for simultaneously tracking the movement of the lips, jaw, tongue, and larynx for human neuroscience research carried out at the bedside. We combined three methods previously used separately: videography to track the lips and jaw, electroglottography to monitor the larynx, and ultrasonography to track the tongue. To characterize this system, we recorded articulator positions and acoustics from six speakers during production of nine American English vowels. We describe processing methods for the extraction of kinematic parameters from the raw signals and methods to account for artifacts across recording conditions. To understand the relationship between kinematics and acoustics, we used regularized linear regression between the vocal tract kinematics and speech acoustics to identify which, and how many, kinematic features are required to explain both across vowel and within vowel acoustics. Furthermore, we used unsupervised matrix factorization to derive "prototypical" articulator shapes, and use them as a basis for articulator analysis. These results demonstrate a multi-modal system to non-invasively monitor speech articulators for clinical human neuroscience applications and introduce novel analytic methods for understanding articulator kinematics.

2:25

1pSCa5. A study of tongue trajectories for English /æ/ using articulatory signals automatically extracted from lingual ultrasound video. Jeff Mielke, Christopher Carignan, and Robin Dodsworth (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, ccarign@ncsu.edu)

While ultrasound imaging has made articulatory phonetics more accessible, quantitative analysis of ultrasound data often reduces speech sounds to tongue contours traced from single video frames, disregarding the temporal aspect of speech. We propose a tracing-free method for directly converting entire ultrasound videos to phonetically interpretable articulatory signals using Principal Component Analysis of image data (Hueber *et al.* 2007). Once a batch of ultrasound images (e.g., 36,000 frames from 10 min at 60 fps) has been reduced to 20 principal components, numerous techniques are available for deriving temporally changing articulatory signals that are both phonetically meaningful and comparable across speakers. Here we apply a regression model to find the linear combination of PCs that is the lingual articulatory analog of the front diagonal of the acoustic vowel space (Z2-Z1). We demonstrate this technique with a study of /æ/ tensing in 20 speakers of North American English varieties with different tensing environments (Labov 2005). Our results show that /m n/ condition a tongue raising gesture that is aligned to the vowel nucleus, while /g/ conditions anticipatory raising toward the velar target. /ŋ/ patterns consistently with the other velar rather than the other nasals.

2:45–3:05 Break

3:05

1pSCa6. Combined analysis of real-time three-dimensional tongue ultrasound and digitized three-dimensional palate impressions: Methods and findings. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

Vocal tract and articulatory imaging has a long and rich history using a wide variety of techniques and equipment. This presentation focuses on combining real-time 3D ultrasound with high-resolution 3D digital scans of palate impressions. Methods for acquiring and analyzing these data will be presented, including efforts to accomplish 3D registration of the tongue and hard palate. Findings from an experiment investigating inter-speaker variability in palate shape and vowel articulation will also be presented.

3:25

1pSCa7. AutoTrace: An automatic system for tracing tongue contours. Gustave V. Hahn-Powell (Linguist, Univ. of Arizona, 2850 N Alvernon Way, Apt. 17, Tucson, AZ 85712, hahnpowell@email.arizona.edu) and Diana Archangeli (Linguist, Univ. of Hong Kong, Tucson, Arizona)

Ultrasound imaging of the tongue is used for analyzing the articulatory features of speech sounds. In order to be able to study the movements of the tongue, the tongue surface contour has to be traced for each recorded image. In order to capture the details of the tongue's movement during speech, the ultrasound video is generally recorded at the highest frame rate available. Detail comes at a price. The number of frames produced from even a single non-trivial experiment is often far too large to trace manually. The Arizona Phonological Imaging Lab (APIL) at the University of Arizona has developed a suite of tools to simplify the labeling and analysis of tongue contours. AutoTrace is a state-of-the-art automatic method for tracing tongue contours that is robust across speakers and languages and operates independently of frame order. The workshop will outline the software installation procedure, introduce the included tools for selecting and preparing training data, provide instructions for automated tracing, and overview a method for measuring the network's accuracy using the Mean Sum of Distances (MSD) metric described by Li *et al.* (2005).

Contributed Papers

3:45

1pSCa8. UATracker: A tool for ultrasound data management. Mohsen Mahdavi Mazdeh and Diana B. Archangeli (Linguist, Univ. of Arizona, 3150 E Bellevue St., #16, Tucson, AZ 85716, mahdavi@email.arizona.edu)

This presentation introduces TraceTracker, a tool for efficiently managing language ultrasound data. Ultrasound imaging of the tongue is used for analyzing the articulatory features of speech sounds. Most analyses involve finding data points from individual images. The number of image frames and the volume of secondary data associated with them tend to grow quickly in speech analysis studies of this type, making it very hard to handle them manually. TraceTracker is a data management tool for organizing, modifying, and performing advanced searches over ultrasound tongue images and the data associated with those images. The setup operation of the program automatically iterates through file systems and generates a comprehensive database containing the image files and information such as the speaker, the video each frame is extracted from, an index, how they have been traced, etc. The program also automatically reads Praat format TextGrid files and associates specific image frames with the corresponding words and speech segments based on the annotations in the grids. Once the database is populated, TraceTracker can be used to tag images, generate copies, and perform advanced search operations over the images based on the aforementioned criteria including the specific sequence of segments in which it lies.

4:00

1pSCa9. Optical flow analysis for measuring tongue-motion. Adriano V. Barbosa (Electron. Eng., Federal Univ. of Minas Gerais, Belo Horizonte, Brazil) and Eric Vatikiotis-Bateson (Linguist, Univ. Br. Columbia, 2613 West Mall, Vancouver, BC V6N2W4, Canada, evb@mail.ubc.ca)

Most attempts to measure motion of the tongue have focused on locating the upper surface of the tongue or specific points on that surface. Recently, we have used our software implementation of optical flow analysis, FlowAnalyzer, to extract measures of tongue motion. The software allows identification of multiple regions of interest, consisting of rectangles whose dimensions and location are user-definable. For example, a large region encompassing the visible tongue body provides general information about the amount and direction (2D) of motion through time; while narrow vertical rectangles can measure the time-varying changes of tongue height at various locations. We will demonstrate the utility of the software, which is freely available upon request to the authors.

4:15

1pSCa10. An acoustic profile of Spanish trill /r/. Ahmed Rivera-Campos and Suzanne E. Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3202, Eden Ave., Cincinnati, OH 45267, riveraam@mail.uc.edu)

Unlike English rhotic, there is limited data on the acoustic profile of Spanish trill /r/. It is well known that one key aspect of the English rhotic /ɹ/ is the lowering of the F3 formant but limited information can be found if Spanish trill shares the same characteristics. Although it has been described that a lowering of F3 is not something that characterizes /r/ production and that F3 values fall under certain ranges that are delimited by vowel contexts, analysis of F3 values has not been done using a large sample of native speakers of Spanish. The following study analyzed the F3 values of 20 participants after production of /r/ by different native speakers of Spanish from different regions of Latin America, and the Caribbean. Analysis of F3 values of /r/ provides information about articulatory requirements for adequate /r/ production. This information will benefit professionals that service individuals with articulatory difficulties or are learning Spanish as a second language.

4:30

1pSCa11. Investigation of the role of the tongue root in Kazakh vowel production using ultrasound. Jonathan N. Washington (Linguist, Indiana Univ., Bloomington, IN 47403-2608, jonwashi@indiana.edu)

It has been argued that Kazakh primarily distinguishes its anterior ("front") vowels from its posterior ("back") vowels through retraction of the tongue root. This analysis is at odds with the traditional assumption that the anteriority of Kazakh vowels is contrasted by tongue body position. The present study uses ultrasound imaging to investigate the extent to which the position of the tongue root and the tongue body are involved in the anteriority contrast in Kazakh. Native speakers of Kazakh were recorded reading words (in carrier sentences) containing target vowels, which were controlled for adjacent consonants and metrical position. An audio recording was also made of these sessions. Frames containing productions of the target vowels were extracted from the ultrasound video and the imaged surface of the tongue was manually traced. Analyses of tongue root and body position were analyzed for each vowel and will be presented together with formant measurements from the audio recordings.

4:45

1pSCa12. Vowel production in sighted children and congenitally blind children. Lucie Menard and Christine Turgeon (Linguist, Universite du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

It is well known that vision plays an important role in speech perception. At the production level, we have recently shown that speakers with congenital visual deprivation produce smaller displacements of the lips (visible articulator) compared to their sighted peers [L. Ménard, C. Toupin, S. Baum, S. Drouin, J. Aubin, and M. Tiede, J. Acoust. Soc. Am. 134, 2975–2987 (2013)]. To further investigate the impact of visual experience on the articulatory gestures used to produce intelligible speech, a speech production study was conducted with blind and sighted school-aged children. Eight congenitally blind children (mean age: 7 years old, from 5 years to 11 years) and eight sighted children (mean age: 7 years old, from 5 years to 11 years) were recorded using a synchronous ultrasound and Optotrak imaging system to record tongue and lip positions. Repetitions of the French vowels /i/, /a/, and /u/ were elicited in a /bVb/ sequence in two prosodic conditions: neutral and under contrastive focus. Tongue contours, lip positions, and formant values were extracted. Acoustic data show that focused syllables are less differentiated from their unfocused counterparts in blind children than in sighted children. Trade-offs between lip and tongue positions are examined.

Session 1pSCb

Speech Communication: Issues in Cross Language and Dialect Perception (Poster Session)

Tessa Bent, Chair
Dept. of Speech and Hearing Sciences, Indiana Univ., Bloomington, IN 47405

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

Contributed Papers

1pSCb1. Cross-language identification of non-native lexical tone. Jennifer Alexander and Yue Wang (Dept. of Linguist, Simon Fraser Univ., 9201 Robert C Brown Hall Bldg., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, jennifer_alexander@sfu.ca)

We extend to lexical-tone systems a model of second-language perception, the Perceptual Assimilation Model (PAM) (Best & Tyler, 2007), to examine whether native-language lexical-tone experience influences identification of novel tone. Native listeners of Cantonese, Thai, Mandarin, and Yoruba hear six CV syllables, each produced with the three phonemic Yoruba tones (High-level/H, Mid-level/M, Low-level/L), presented randomly three times. In a 3-AFC task, participants indicate a syllable's tone by selecting from a set of arrows the one that illustrates its pitch trajectory. Accuracy scores (proportion correct) were submitted to a two-way rANOVA with L1-Group (x4) as the between-subjects factor and Tone (x3) as the within-subjects factor. There was no main effect of Tone or Group. The Tone-by-Group interaction was significant ($p = 0.031$) but driven by one group: Thai listeners identified H and M more accurately than L (both $p < 0.05$), though L accuracy was above chance (59%; chance = 33.33%). Tone-error patterns indicate that Thai listeners primarily confused L with M (two-way L1-Group x Response-pattern rANOVA $p < 0.05$). Overall, despite their different tonal-L1 backgrounds, listeners performed comparably. As predicted by the PAM, listeners attended to gradient phonetic detail and acoustic cues relevant to L1 phoneme distinctions (F0 height/direction) in order to classify non-native contrasts. [NSF grant #0965227.]

1pSCb2. Spectral and duration cues of English vowel identification for Chinese-native listeners. Sha Tao, Lin Mi, Wenjing Wang, Qi Dong (Cognit. Neurosci. and Learning, Beijing Normal Univ., State Key Lab for Cognit. Neurosci. and Learning, Beijing Normal University, Beijing 100875, China, taosha@bnu.edu.cn), and Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

This study was to investigate how Chinese-native listeners use spectral and duration cues for English vowel identification. The first experiment was to examine whether Chinese-native listeners' English vowel perception was related to their sensitivity to the change of vowel formant frequency that is a critical spectral cue to vowel identification. Identification of 12 isolated American English vowels was measured for 52 Chinese college students in Beijing. Thresholds of vowel formant discrimination were also examined for these students. Results showed that there was a significantly moderate correlation between Chinese college students' English vowel identification and their thresholds of vowel formant discrimination. That is, the lower vowel formant threshold of listeners, the better vowel identification. However, the moderate correlation between vowel identification and formant discrimination suggested some other factors accounting for the individual variability in English vowel identification for Chinese-native listeners. In Experiment 2, vowel identification was measured with and without duration cues, showing that vowel identification was reduced by 5.1% when duration cue was removed. Further analysis suggested that for the listeners who depended less on duration cue, the better thresholds of formant discrimination, the higher scores of vowel identification, but no such correlation for listeners who used duration cues remarkably.

1pSCb3. The influence of lexical status in the perception of English allophones by Korean learners. Kyung-Ho Kim and Jeong-Im Han (English, Konkuk Univ., 120 Neungdong-ro, Gwangjin-gu, Seoul 143-701, South Korea, gabrieltotti88@gmail.com)

This study investigated whether the allophonic contrast in the second language (L2) may require contact with the lexicon to influence the perception. Given that English medial voiceless stops occur with aspiration in stressed, but without aspiration in unstressed syllables, Korean learners of English were tested for aspirated and unaspirated allophones of /p/ for perceptual preference in appropriate and inappropriate stress contexts in the second syllable of disyllabic words. The stimuli included four types of non-words and eight pairs of real words (four pairs each for high-frequency and low-frequency words), and participants were asked to judge the perceptual preference of each token on a 7-scale (1 = a bad example, 7 = a good example). The results demonstrated that in tests with non-words, there was no significant difference in the ratings as a function of context appropriateness (e.g., [ípɚ] vs. [íphɚ]), with higher rankings for initially-stressed words. By contrast, in real words, participants preferred the correct allophones (e.g., [képɚ] vs. [képhɚ] "caper"). The frequency of real words further showed a significant effect. This finding suggests that allophony in L2 is driven by lexicality (Whalen *et al.*, 1997). Exemplar theory (Pierrehumbert 2001, 2002) provides a more effective means of modeling this finding than do traditional approaches.

1pSCb4. The perception of English coda obstruents by Mandarin and Korean second language learners. Yen-Chen Hao (Modern Foreign Lang. and Literatures, Univ. of Tennessee, 510 14th St. #508, Knoxville, TN 37916, yenchenhao@gmail.com) and Kenneth de Jong (Linguist, Indiana Univ., Bloomington, IN)

This study investigates the perception of English obstruents by learners whose native language is either Mandarin, which does not permit coda obstruents, or Korean, which neutralizes laryngeal and manner contrasts into voiceless stop codas. The stimuli are native productions of eight English obstruents /p b t d f v θ ð/ combined with the vowel /ɑ/ in different prosodic contexts. Forty-one Mandarin and 40 Korean speakers identified the consonant from the auditorily presented stimuli. The results show that the two groups do not differ in their accuracy in the onset position, indicating that they are comparable in their proficiency. However, the Mandarin speakers are more accurate in the coda position than the Koreans. When the fricatives and stops are analyzed separately, it shows that the two groups do not differ with fricatives, yet

the Mandarin speakers are more accurate than the Koreans with stops. These findings suggest that having stop codas in their L1 does not necessarily facilitate Koreans' acquisition of the L2 sounds. Despite their L1 differences, the two groups display very similar perceptual biases in their error patterns. However, not all of them can be explained by L1 transfer or universal markedness, suggesting other language-independent factors in L2 perception.

1pSCb5. Effect of phonetic training on the perception of English consonants by Greek speakers in quiet and noise conditions. Angelos Lengeris and Katerina Nicolaidis (Theor. and Appl. Linguist, Aristotle Univ. of Thessaloniki, School of English, Aristotle University, Thessaloniki 541 24, Greece, lengeris@enl.auth.gr)

The present study employed high-variability phonetic training (multiple words spoken by multiple talkers) to improve the identification of English consonants by native speakers of Greek. The trainees completed five sessions of identification training with feedback for seven English consonants (contrasting voiced vs. voiceless stops and alveolar vs. postalveolar fricatives) each consisting of 198 trials with a different English speaker in each session. Another group of Greek speakers served as controls, i.e., completed the pre/post test but received no training. Pre/post tests included English consonant identification in quiet and noise. In the noise condition, participants identified consonants in the presence of a competing English speaker at a signal-to-noise ratio of -2dB. The results showed that training significantly improved English consonant perception for the group that received training but not for the control group in both quiet and noise. The results add to the existing evidence that supports the effectiveness of the high-variability approach to second-language segmental training.

1pSCb6. Perceptual warping of phonetic space applies beyond known phonetic categories: Evidence from the perceptual magnet effect. Bozena Pajak (Brain & Cognit. Sci., Univ. of Rochester, 1735 N Paulina St. Apt. 509, Chicago, Illinois 60622, bpajak@bcs.rochester.edu), Page Piccinini, and Roger Levy (Linguist, Univ. of California, San Diego, San Diego, CA)

What is the mental representation of phonetic space? Perceptual reorganization in infancy yields a reconfigured space “warped” around native-language (L1) categories. Is this reconfiguration entirely specific to L1 category inventory? Or does it apply to a broader range of category distinctions that are non-native, yet discriminable due to being defined by phonetic dimensions informative in the listener's L1 (Bohn & Best, 2012; Pajak, 2012)? Here we address this question by studying perceptual magnets, which involve attrition of within-category distinctions and enhancement of distinctions across category boundaries (Kuhl, 1991). We focus on segmental length, known to yield L1-specific perceptual magnets: e.g., L1-Finnish listeners have one for [t]/[tt], but L1-Dutch listeners, who lack (exclusively) length-based contrasts, do not (Herren & Schouten, 2008). We tested 31 L1-Korean listeners in an AX discrimination task for [n]-[nn] and [f]-[ff] continua. Korean listeners have been shown to discriminate both (Pajak, 2012), despite only having the former set in the inventory. We found perceptual magnets for both continua, demonstrating that perceptual warping goes beyond the specific L1 categories: when a phonetic dimension is informative for contrasting some L1 categories, perceptual warping applies not only to the tokens from those categories, but also to that dimension more generally.

1pSCb7. Language mode effects on second language categorical perception. Beatriz Lopez Prego and Allard Jongman (Linguist, Univ. of Kansas, 1145 Pennsylvania St., Lawrence, KS 66044, lopezb@ku.edu)

This study investigates the perception of the /b/-/p/ voicing contrast in English and Spanish by native English listeners, native Spanish listeners, and highly proficient Spanish-speaking second-language (L2) learners of English with a late onset of acquisition (mean = 10.8) and at least three-year residence in an English-speaking environment. Participants completed a forced-choice identification task where they identified target syllables in a Voice Onset Time (VOT) continuum as "pi" or "bi." They listened to 10 blocks of 19 equidistant steps ranging from + 88 ms-VOT to -89 ms-VOT. Between blocks, subjects read and wrote responses to language background questions, thus actively processing the target language. Monolinguals completed the task in their native language (L1). L2 learners completed the task once in their L1 and once in their L2, thus providing a manipulation of "language mode" (Grosjean, 2001). The results showed that L2 learners' category boundary in English did not differ from that of monolingual English listeners, but their category boundary in Spanish differed from that of monolingual Spanish listeners and from their own category boundary in English. These results suggest that the language mode manipulation was successful and that L2 learners can develop new phonetic categories, but this may have an impact on their L1 categories.

1pSCb8. Processing of English-accented Spanish voice onset time by Spanish speakers with low English experience. Fernando Llanos (School of Lang. and Cultures, Purdue Univ., Stanley Coulter Hall, 640 Oval Dr., West Lafayette, IN 47907, fllanos@purdue.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Previous research (Llanos & Francis, 2014) shows that the processing of foreign accented speech sounds can be affected by listeners' familiarity with the language that causes the accent. Highly familiar listeners treat foreign accented sounds as foreign sounds while less familiar listeners treat them natively. The present study tests the hypothesis that less familiar listeners may nevertheless be able to apply foreign categorization patterns to accented words by recalibrating phonetic expectations according to acoustic information provided by immediate phonetic context. Two groups of Spanish native speakers with little English experience will identify tokens drawn from a digitally edited VOT continuum ranging from baso "glass" (-60 ms VOT) to paso "step" (60 ms VOT). Tokens are embedded in a series of Spanish words beginning with /b/ and /p/ to provide phonetic context. In the English-accented condition, context words are digitally modified to exhibit English-like VOT values for /b/ (10 ms) and /p/ (60 ms). In the Spanish condition, these tokens are edited to exhibit prototypical Spanish /b/ (-90 ms) and /p/ (10 ms) VOT values. If listeners can accommodate foreign accented sounds according to expectations provided by immediate phonetic context, then listeners' VOT boundary in the English-accented condition should be significantly higher than in the Spanish condition.

1pSCb9. Amount of exposure and its effect on perception of second language front vowels in English. Andrew Jeske (Linguist, Univ. of Pittsburgh, 3211 Brereton St., Pittsburgh, PA 15219, arjeske@gmail.com)

Experience with a second language (L2) has been shown to positively affect learners' perception of L2 sounds. However, few studies have focused on how the amount of L2 exposure in foreign language classrooms impacts perception of L2 sounds during the incipient stages of language learning in school-age children. To determine what effect, if any, the amount of L2 exposure has on perception, 64 students from a Spanish-English bilingual elementary school and 60 students from two non-bilingual elementary schools participated in an AX Categorical Discrimination task, which contained tokens of five English front vowels: /i ɪ e ɛ æ/. Results show that students from the bilingual school earned perception scores significantly higher than those earned by the students from the non-bilingual school ($p = 0.002$). However, an ANOVA found there to be no significant simple main effect for grade or significant correlation between grade level and school type. The bilingual school students perceived all within-category word pairings (e.g., bat-bat) significantly more accurately than the non-bilingual school students suggesting that increased, early exposure to an L2 may heighten one's ability to disregard irrelevant, interpersonal phonetic differences and lead to a within-category perceptual advantage over those with less L2 exposure early on.

1pSCb10. Does second language experience modulate perception of tones in a third language? Zhen Qin and Allard Jongman (Linguist, Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045, qinzhenquentin2@ku.edu)

Previous studies have shown that English speakers pay attention to pitch height rather than direction, whereas Mandarin speakers are more sensitive to pitch direction than height in perception of lexical tones. The present study addresses if a second language (L2, i.e., Mandarin) overrides the influence of a native language (L1, i.e., English) in modulating listeners' use of pitch cues in the perception of tones in a third language (L3, i.e., Cantonese). English-speaking L2 learners (L2ers) of Mandarin constituted the

target group. Mandarin speakers and English speakers without knowledge of Mandarin were included as control groups. In Experiment 1, all groups, naïve to Cantonese tones, discriminated Cantonese tones by distinguishing either a contour tone from a level tone (pitch direction pair) or a level tone from another level tone (pitch height pair). The results showed that L2ers patterned differently from both control groups with regard to pitch cues under the influence of L2 experience. The acoustics of the tones also affected all listeners' discrimination. In Experiment 2, L2ers were instructed to identify Mandarin tones to measure their sensitivity to L2 tones. The results showed that L2ers' sensitivity to L2 tones is not necessarily correlated with their perception of L3 tones.

1pSCb11. Does early foreign language learning in school affect phonemic discrimination in adulthood? Tetsuo Harada (School of Education, Waseda Univ., 1-6-1 Nishi Waseda, Shinjuku, Tokyo 169-8050, Japan, tharada@waseda.jp)

Long-term effects of early foreign language learning with a few hours' classroom contact per week on speech perception are controversial: some studies show age effects of minimal English input in childhood on phonemic perception in adulthood, but others don't (e.g., Lin *et al.*, 2004). This study investigated effects of a younger starting age in a situation of minimal exposure on perception of English consonants under noise conditions. The listeners were two groups of Japanese university students: early learners ($n = 21$) who started studying English in kindergarten or elementary school, and late learners ($n = 24$) who began to study in junior high school. The selected target phonemes were word-medial approximants (/l, r/). Each nonword (i.e., *ala*, *ara*), produced by six native talkers, was combined with speech babble at the signal-to-noise ratios (SNRs) of 8 dB (medium noise) and 0 dB (quite high noise for L2 listeners). A discrimination test was given in the ABX format. Results showed that the late learners discriminated /l/ and /r/ better than the early learners regardless of the noise conditions and talker differences ($p < 0.05$). A multiple regression analysis revealed that length of learning and English use could contribute to their discrimination ability.

1pSCb12. The identification of American English vowels by native speakers of Japanese before three nasal consonants. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp)

Native speakers of Japanese identified American English vowels that are uttered before three nasal consonants /m, n, ŋ/ and three oral stop consonants /b, d, g/. Of the seven vowels /i, ɪ, eɪ, ɛ, æ, ɑ, ʌ/, /æ/ was generally less accurately identified before nasal consonants than before oral stop consonants, and this tendency was stronger when /ŋ/ follows. This tendency is probably attributed to the extended raising of /æ/ before /ŋ/ and the Japanese listeners' limited sensitivity to differentiate three nasal phonemes in coda position. /ɪ/, on the other hand, was identified more correctly before /ŋ/ than before the other two nasal consonants, also probably because the vowel is raised before /ŋ/. This vowel was more often misidentified as /ɛ/ before /m/ and /n/. /ɑ/ and /ʌ/ were less accurately identified before stop consonants, but after nasal consonants, /ʌ/ was more often misidentified as /ɑ/. /ɑ/ and /ʌ/ may sound alike to Japanese listeners in every context, but before nasal contexts, both of these vowels may sound closer to the Japanese vowel /o/. The results generally revealed that identification accuracy cannot be solely accounted for in terms of the place of articulation of the following consonant.

1pSCb13. Effects of beliefs about first language orthography on second language vowel perception. Mara Haslam (Dept. of Lang. Education, Stockholm Univ., S:t Ansgars väg 4, Solna 16951, Sweden, mara.haslam@gmail.com)

Recent research has identified that L1 orthography can affect perception of vowels in a second language (e.g., Escudero and Wanrooij, 2010). The present study investigates the effect that participants' beliefs about orthography have on their ability to perceive vowels in a second language. English- and Polish-speaking learners of Swedish have to encounter some new vowel sounds and also the characters that are used to represent them, e.g., å, ä, and ö. New survey data of native speakers of English, Polish, and Swedish confirm that L1 English speakers see these characters like these as familiar letters with diacritics, while L1 Swedish and L1 Polish speakers tend to see these types of characters as different characters of the alphabet. These differing beliefs about orthography may cause English speakers to confuse the vowels represented in Swedish by the characters å, ä and ö with vowels represented by the characters a, a, and o, respectively, while Polish speakers would not be similarly affected. Results of a Swedish vowel perception study conducted with native speakers of English and Polish after exposure to Swedish words containing these characters will be presented. These results contribute to increasing knowledge about the relationship between L1 orthography and L2 phonology.

1pSCb14. A preliminary investigation of the effect of dialect on the perception of Korean sibilant fricatives. Jeffrey J. Holliday (Second Lang. Studies, Indiana Univ., 1021 E. Third. St., Memorial Hall M03, Bloomington, IN 47405, jjhollid@indiana.edu) and Hyunjung Lee (English, Hankyong National Univ., Anseong, Gyeonggi-do, South Korea)

Korean has two sibilant fricatives, /s^h/ and /s*/, that are phonologically contrastive in the Seoul dialect but are widely believed to be phonetically neutralized in the Gyeongsang dialects spoken in southeastern South Korea, with both fricatives being acoustically realized as [s^h]. The current study investigated the degree to which the perception of these fricatives by Seoul listeners is affected by knowledge of the speaker's dialect. In the first task, the stimuli were two fricative-initial minimal pairs (i.e., four words) produced by 20 speakers each from Seoul and Gyeongsang. Half of the 18 listeners were told that the speakers were from Seoul, and the other half were told they were from Gyeongsang. Listeners identified the 160 word-initial fricatives and provided a goodness rating for each. It was found that neither the speaker's actual dialect nor the primed dialect had a significant effect on either identification accuracy or listeners' goodness ratings. In a second task, listeners identified tokens from a seven-step continuum from [sada] to [s*ada]. It was found that listeners who were primed for Gyeongsang dialect were more likely to perceive tokens as /s*/ than listeners primed for Seoul, which may reflect a dialect-based hypercorrective perceptual bias.

1pSCb15. Language is not destiny: Task-specific factors, and not just native language perceptual biases, influence foreign sound categorization strategies. Jessamyn L. Schertz and Andrew Lotto (Univ. of Arizona, Douglass 200, Tucson, AZ 85721, jschertz@email.arizona.edu)

Listeners were trained to distinguish two novel classes of speech sounds differing in both Voice Onset Time (VOT) and fundamental frequency at vowel onset (f0). One group was shown Korean orthography during the training period ("symbols" group) and the other English orthography ("letters" group). During a subsequent test phase, listeners classified sounds with mismatched VOT and f0. The two groups relied on different cues to categorize the contrast: those exposed to symbols used f0, while those exposed to letters used VOT. A second experiment employed the same paradigm, but the two dimensions defining the contrast were closure duration (instead of f0) and VOT. In this more difficult experiment, successful listeners in the "letters" group again classified the hybrid stimuli based on VOT, while the single listener in the "symbols" group who passed the learning criterion used closure duration. In both experiments, subjects showed different categorization patterns based on orthography used in the presentation, even though orthography was irrelevant for the experimental task. Listeners relied on VOT when the stimuli were presented with English, but not foreign, orthography, showing that task-related information (as opposed to native language biases alone) can direct attention to different acoustic cues in foreign contrast classification.

1pSCb16. Generational difference in the perception of high-toned [il] in Seoul Korean. Sunghye Cho (Univ. of Pennsylvania, 3514 Lancaster Ave., Apt. 106, Philadelphia, PA 19104, csunghye@sas.upenn.edu)

A word-initial [il] is most frequently H-toned in Seoul Korean (SK) when it means one, out of three homophones, one, day, and work (Jun & Cha, 2011). However, Cho (2014) finds that 25% of teenagers always produce [il] with a H tone, regardless of its meaning. This paper examines how young SK speakers perceive the phenomenon. Thirty-seven SK speakers (aged 14–29) participated in two identification tasks, hearing only [il] in the first task and four [il]-initial minimal pairs in the second task. All target words were manipulated into five pitch levels with 30 Hz intervals. In the first task, the 20s group identified [il] as one 70% of the time at higher pitch levels, while the teenagers identified [il] as one about 50% of the time at all pitch levels. In the second task, the 20s group showed a categorical perception, identifying [il]-initial words as one only at higher pitch levels, while the teenagers did not. The results suggest that the teenagers are aware that some peers always produce [il] with a H tone. It explains that the 20s group could identify the meanings of [il] depending on the pitch, while the teenagers could not.

1pSCb17. The effect of perceived talker race on phonetic imitation of pin-pen words. Qingyang Yan (Linguist, The Ohio State Univ., 591 Harley Dr., Apt. 10, Columbus, OH 43212, yan@ling.ohio-state.edu)

The current study investigated the phonetic imitation of the PIN-PEN merger by nonmerged participants. An auditory shadowing task was used to examine how participants changed their /ɪ/ and /ɛ/ productions after auditory exposure to merged and nonmerged voices. Black and white talker photos were used as visual cues to talker race. The pairing of voices (merged and nonmerged) with the talker photos (black and white) was counterbalanced across participants. A third group of participants completed the task without talker photos. Participants' explicit talker attitudes were assessed by a questionnaire, and their implicit racial attitudes were measured by an Implicit Association Task. Nonmerged participants imitated the PIN-PEN merger, and the degree of imitation varied depending on the experimental condition. The merged voice elicited more imitation when it was presented without a talker photo or with the black talker photo than with the white talker photo. No effect of explicit talker attitudes or implicit racial attitudes on the degree of imitation was observed. These results suggest that phonetic imitation of the PIN-PEN merger is more complex than an automatic response to the merged voice and that it is mediated by perceived talker race.

1pSCb18. Foreign-accent discrimination with words and sentences. Eriko Atagi (Volen National Ctr. for Complex Systems, Brandeis Univ., Volen National Ctr. for Complex Systems, MS 013, Brandeis University, 415 South St., Waltham, MA 02454-9110, eatagi@brandeis.edu) and Tessa Bent (Dept. of Speech & Hearing Sci., Indiana Univ., Bloomington, IN)

Native listeners can detect a foreign accent in very short stimuli; however, foreign-accent detection is more accurate with longer stimuli (Park, 2008; Flege, 1984). The current study investigated native listeners' sensitivity to the characteristics that differentiate between accents—both foreign versus native accents and one foreign accent versus another—in words and sentences. Listeners heard pairs of talkers reading the same word or sentence and indicated whether the talkers had the same or different native language backgrounds. Talkers included two native talkers (Midland dialect) and six nonnative talkers from three native language backgrounds (German, Mandarin, and Korean). Sensitivity varied significantly depending on the specific accent pairings and stimulus type. Listeners were most sensitive when the talker pair included a native talker, but could detect the difference between two nonnative accents. Furthermore, listeners were generally more sensitive with sentences than with words. However, for one nonnative pairing, listeners exhibited higher sensitivity with words; for another, listeners' sensitivity did not differ significantly across stimulus types. These results suggest that accent discrimination is not simply influenced by stimulus length. Sentences may provide listeners with opportunities to perceive similarities between nonnative talkers, which are not salient in single words. [Work supported by NIDCD T32 DC00012.]

1pSCb19. Stimulus length and scale label effects on the acoustic correlates of foreign accent ratings. Elizabeth A. McCullough (Linguist, Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, eam@ling.ohio-state.edu)

Previous studies have investigated acoustic correlates of accentedness ratings, but methodological differences make it difficult to compare their results directly. The present experiment investigated how choices about stimulus length and rating scale labels influence the acoustic correlates of listeners' rating responses. Four conditions crossed two stimulus lengths (CV syllable vs. disyllabic word) with two sets of rating labels ("no foreign accent"/"strong foreign accent" vs. "native"/"not native"). Monolingual American English listeners heard samples of English from native speakers of American English, Hindi, Korean, Mandarin, and Spanish and indicated their responses on a continuous rating line. Regression models evaluated the correlations between listeners' ratings and a variety of acoustic properties. Patterns for accentedness and non-nativeness ratings were identical. VOT, F1, and F2 correlated with ratings on all stimuli, but vowel duration correlated with ratings on disyllabic word stimuli only. If vowel duration is interpreted as a reflection of global temporal properties, this result suggests that listeners may perceive such properties in utterances as short as two syllables. Thus, stimulus design is vital in identifying components of foreign accent perception that are related to differences between a talker's first and second languages as opposed to components that are related to general fluency.

1pSCb20. Language proficiency, context influence foreign-accent adaptation. Cynthia P. Blanco (Linguist, Univ. of Texas at Austin, 305 E. 23rd St., Austin, TX 78712, cindyblanco@utexas.edu), Hoyoung Yi (Commun. Sci. & Disord., Univ. of Texas at Austin, Austin, TX), Elisa Ferracane, and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

Listeners adapt quickly to changes in accent (Bradlow & Bent, 2003; Clarke & Garrett, 2004; inter alia). The cause of this brief delay may be due to the cost of processing accented speech, or may reflect a surprise effect associated with task expectations (Floccia *et al.*, 2009). The present study examines a link between accent familiarity and processing delays with listeners who have varying degrees of familiarity with target languages: monolingual Texans with little or no formal exposure to Spanish, early Spanish-English bilinguals, and Korean learners of English. Participants heard four blocks of English sentences—Blocks 1 and 4 were produced by two native speakers of American English, and Blocks 2 and 3 were produced by native speakers of Spanish or Korean- and responded to written probe words. All listener groups responded more slowly after an accent change; however, the degree of delay varied with language proficiency. L1 Korean listeners were less delayed by Korean-accented speech than the other listeners, while changes to Spanish-accented speech were processed most slowly by Spanish-English bilinguals. The results suggest that adaptation to foreign-accented speech depends on language familiarity and task expectations. The processing delays are analyzed in light of intelligibility and accentedness measures.

1pSCb21. When two become one—Orthography helps link two free variants to one lexical entry. Chung-Lin Yang (Linguist, Indiana Univ.- Bloomington, Memorial Hall 322, 1021 E 3rd St., Bloomington, IN 47408, cy1@indiana.edu) and Isabelle Darcy (Second Lang. Studies, Indiana Univ.- Bloomington, Bloomington, IN)

L2 learners can become better at distinguishing an unfamiliar contrast by knowing the corresponding orthographic forms (e.g., Escudero *et al.*, 2008). We ask whether learners could associate two free variants with the same lexical entry when the orthographic form was provided during learning. American learners learned an artificial language where [p]-[b] were in free variation (both were spelled as <p>) (test condition) while [t]-[d] were contrastive (control condition), or vice-versa ([t]-[d] in test, counterbalanced across subjects). Using a word-learning paradigm modified from Hayes-Harb *et al.* (2010), in the learning phase, participants heard novel words paired with pictures. One subgroup of learners saw the spellings as well ("Orth+"), while another did not (i.e., auditory only, "Orth−"). Then in a picture-auditory word matching task, the new form of the word was paired with the original picture. Orth+ learners were expected to be more accurate at accepting the variant as the correct label for the original test item than Orth−. The results showed that Orth+ learners detected and learned the [p]-[b] free variation significantly better than Orth− ($p < 0.05$), but not the [t]-[d] free variation. Thus, the benefit of orthography in speech learning could vary depending on the specific contrasts at hand.

Session 1pUW

Underwater Acoustics: Understanding the Target/Waveguide System–Measurement and Modeling II

Aubrey L. Espana, Chair
Acoustics Dept., Applied Physics Lab, Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105

Chair's Introduction—1:25

Invited Paper

1:30

1pUW1. Mapping bistatic scattering from spherical and cylindrical targets using an autonomous underwater vehicle in BAYEX'14 experiment. Erin M. Fischell, Stephanie Petillo, Thomas Howe, and Henrik Schmidt (Mech. Eng., MIT, 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu)

In May 2014, the MIT Laboratory for Autonomous Marine Sensing Systems (LAMSS) participated in the BAYEX'14 experiment with the goal of collecting full bistatic data sets around proud spherical and cylindrical targets for use in real-time autonomous target localization and classification. The BAYEX source was set to insonify both targets, and was triggered to ping at the start of each second using GPS PPS. The MIT Bluefin 21 in. AUV Unicorn, fitted with a 16-element nose array, was deployed in broadside sampling behaviors to collect the bistatic scattered data set. The AUV's Chip Scale Atomic Clock was synchronized to GPS on the surface, and the data was logged using a PPS triggered analog to digital conversion system to ensure synchronization with the source. The MIT LAMSS operational paradigm allowed the vehicle to be unpacked, tested and deployed over the brief three-day interval available for operations. MOOS-IvP and acoustic communication enabled the group to command AUV mission changes in situ based on data collection needs. During data collection, the vehicle demonstrated real-time signal processing and target localization, and the bistatic datasets were used to demonstrate real-time target classification in simulation. [Work supported by ONR Code 322OA.]

Contributed Papers

1:50

1pUW2. Elastic features visible on canonical targets with high frequency imaging during the 2014 St. Andrews Bay experiments. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu), Timothy M. Marston, Steven G. Kargl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Daniel S. Plotnick (Phys. and Astronomy, Washington State Univ., Pullman, WA), Aubrey Espana, and Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

During the 2014 St. Andrews Bay experiments some canonical metallic targets (a hollow sphere and some circular cylinders) were viewed with a synthetic aperture sonar (SAS) capable of acquiring data using a 110–190 kHz chirped source. The targets rested on mud-covered sand and were typically at a range of 20 m. Fast reversible SAS processing using an extension of line-scan quasi-holography [K. Baik, C. Dudley, and P. L. Marston, J. Acoust. Soc. Am. 130, 3838–3851 (2011)] was used to extract relevant signal content from images. The significance of target elastic responses in extracted signals was evident from the frequency response and/or the time-domain response. For example, the negative group velocity guided wave enhancement of the backscattering by the sphere was clearly visible near 180 kHz. [For a ray model of this type of enhancement see: G. Kaduchak, D. H. Hughes, and P. L. Marston, J. Acoust. Soc. Am. 96, 3704–3714 (1994).] In another example, the timing of a sequence of near broadside echoes from a solid aluminum cylinder was consistent with reflection and internal reverberation of elastic waves. These observations support the value of combining reversible imaging with models interpreted using rays. [Work supported by ONR and SERDP.]

2:05

1pUW3. Boundary enhanced coupling processes for rotated horizontal solid aluminum cylinders: Helical rays, synthetic aperture sonar images, and coupling conditions. Jon R. La Follett (Shell International Exploration and Production Inc., Houston, TX) and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Experiments with solid aluminum cylinders placed near a flat free surface provide insight into scattering processes relevant to other flat reflecting boundaries [J. R. La Follett, K. L. Williams, and P. L. Marston, J. Acoust. Soc. Am. 130, 669–672 (2011); J. R. La Follett, Ph.D. thesis, WSU (2010)]. This presentation concerns the coupling to surface guided leaky Rayleigh waves that have been shown to contribute significantly to backscattering by solid metallic cylinders [K. Gipson and P. L. Marston, J. Acoust. Soc. Am. 106, 1673–1689 (1999)]. The emphasis here is on horizontal cylinders rotated about a vertical axis away from broadside viewed at grazing incidence. The range of rotation angles for which helical rays can contribute is limited in the free field by the cylinder's length [F. J. Blonigen and P. L. Marston, J. Acoust. Soc. Am. 112, 528–536 (2002)]. Some examples of surface enhanced backscattering may be summarized as follows. In agreement with geometrical considerations, the angular range for coupling to helical rays may be significantly extended when a short cylinder is adjacent to a flat surface. In addition, the presence of a flat surface splits synthetic aperture sonar (SAS) image features from various guided wave mechanisms on rotated cylinders. [Work supported by ONR.]

Invited Papers

2:20

1pUW4. Denoising structural echoes of elastic targets using spatial time–frequency distributions. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Structural echoes of underwater elastic targets, used for detection and classification purposes, can be highly localized in the time–frequency domain and can be aspect-dependent. Hence, such structural echoes recorded along a distributed (synthetic) aperture, e.g., using a moving receiver platform, would not meet the stationarity and multiple snapshots requirements of common subspace array processing methods used for denoising array data based on their estimated covariance matrix. To handle these scenarios, a generalized space–time–frequency covariance matrix can be computed from the single-snapshot data using Cohen's class time-frequency distributions between all sensor data pairs. This space–time–frequency covariance matrix automatically accounts for the inherent coherence across the time-frequency plane of the received nonstationary echoes emanating from the same target. Hence, identifying the signal's subspace from the eigenstructure of this space–time–frequency covariance matrix provides a means for denoising these non-stationary structural echoes by spreading the clutter and noise power in the time–frequency domain. The performance of the proposed methodology will be demonstrated using numerical simulations and at-sea data.

2:40

1pUW5. Measurements and modeling of acoustic scattering from targets in littoral environments. Harry J. Simpson (Physical Acoust. Branch, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, VA20375, harry.simpson@nrl.navy.mil), Zackary J. Waters, Timothy J. Yoder, Brian H. Houston (Physical Acoust. Branch, Naval Res. Lab., Washington, DC), Kyrie K. Jig, Roger R. Volk (Sotera Defense Solution, Crofton, MD), and Joseph A. Bucaro (Excet, Inc., Springfield, VA)

Broadband laboratory and at-sea measurements systems have been built by NRL to quantify the acoustic target strength of objects sitting on or in the bottom of littoral environments. Over the past decade, these measurements and the subsequent modeling of the target strength have helped to develop an understanding of how the environment, especially near the bottom interface, impacts the structural acoustic response of a variety of objects. In this talk we will present a set of laboratory, at-sea rail and AUV based back scatter, forward scatter, and propagation measurements with subsequent analysis to understand the impact of the littoral environment. Simple targets such as spheres, along with UXO targets will be discussed. The analysis will be focused on quantifying the changes to target strength as a result of being near the bottom interface. In addition to the traditional backscatter or monosatic target strength, we focus upon efforts to investigate the multi-static scattering from targets. [Work supported by ONR.]

3:00–3:15 Break

Contributed Papers

3:15

1pUW6. TREX13 target experiments and case study: Comparison of aluminum cylinder data to combined finite element/physical acoustics modeling. Kevin Williams, Steven G. Kargl, and Aubrey L. Espana (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, williams@apl.washington.edu)

The apparatus and experimental procedure used during the target portion of TREX13 are described. A primary goal of the TREX13 target experiments was to test the high speed modeling methods developed and previously tested as part of efforts in more controlled environments where the sediment/water interface was flat. At issue is to what extent the simplified physics used in our models can predict the changes seen in acoustic templates (target strength versus angle and frequency) as a function of grazing angle, i.e., the Target-In-the-Environment-Response (TIER), for a target proud on an unprepared "natural" sand sediment interface. Data/model comparisons for a 3 ft. long, 1 ft. diameter cylinder are used as a case study. These comparisons indicate that much of the general TIER dependence is indeed captured and allows one to understand/predict geometries where the broadest band of TIER information can be obtained. This case study indicates the predictive utility of dissecting the target physics at the expense of making the model results "inexact" from a purely finite element, constitutive equation standpoint. [Work supported by ONR and SERDP.]

3:30

1pUW7. Predicting the acoustic response of complicated targets in complicated environments using a hybrid finite element/propagation model. Aubrey L. Espana, Kevin L. Williams, Steven G. Kargl (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, aespana@apl.washington.edu), Marten J. Nijhof (Acoust. and Sonar, TNO, Den Haag, Netherlands), Daniel S. Plotnick, and Philip L. Marston (Phys. and Astronomy, Washington State Univ., Pullman, WA)

Previous work has shown that hybrid finite element (FE)/propagation models are a viable tool for estimating the Target-In-The-Environment-Response, or TIER, for simple shapes such as cylinders and pipes on a flat, undisturbed sand/water interface [K. L. Williams *et al*., J. Acoust. Soc. Am 127, 3356–3371 (2010)]. Here we examine their use for more complicated targets located in complicated ocean environments. The targets examined include various munitions and ordnance-like targets, with intricate internal structure and filled with either air or water. A hybrid FE/propagation model is used to predict their TIER on flat, undisturbed sand. Data acquired during the target portion of TREX13 is used to validate the model results. Next, the target response is investigated in a more complicated environment, being partially buried with their axis tilted w.r.t. the flat sand interface. Again model results are validated using TREX13 data, as well as data acquired in a controlled tank experiment. These comparisons highlight the feasibility of using hybrid models for complex target/environment configurations, as well possible limitations due to the effects of multiple scattering.

3:45

1pUW8. A correlation analysis of the Naval Surface Warfare Center Panama City Division's (NSWC PCD) database of simulated and collected target scattering responses focused on automated target recognition. Raymond Lim, David E. Malphurs, James L. Prater, Kwang H. Lee, and Gary S. Sammelmann (Code X11, NSWC Panama City Div., 110 Vernon Ave, Code X11, Panama City, FL 32407-7001, raymond.lim@navy.mil)

Recently, NSWC PCD participated in a number of computational and experimental efforts aimed at assembling a database of sonar scattering responses encompassing a variety of objects including UXO, cylindrical shapes, and other clutter-type objects. The range of data available on these objects consists of a simulated component generated with 3D finite element calculations coupled to a fast Helmholtz-equation-based propagation scheme, a well-controlled experimental component collected in NSWC PCD's pond facilities, and a component of measurements in realistic underwater environments off Panama City, FL (TREX13 and BayEX14). The goal is to use the database to test schemes for automating reliable separation of these objects into desired classes. Here, we report on an initial correlation analysis of the database projected onto the target aspect vs frequency plane to assess the feasibility of the simulated component against the measured ones, to investigate some basic questions regarding environmental and range effects on class separation, and to try and identify phenomena in this plane useful for classification. [Work supported by ONR and SERDP.]

4:05

1pUW9. Identifying buried unexploded ordnance with structural acoustics based numerically trained classifiers: Laboratory demonstrations. Zachary J. Waters, Harry J. Simpson, Brian H. Houston (Physical Acoust. - Code 7130, Naval Res. Lab., 4555 Overlook Ave. SW, Bldg 2. Rm. 186, Washington, DC 20375, zachary.waters@nrl.navy.mil), Kyrie Jig, Roger Volk, Timothy J. Yoder (Sotera Defense Solutions Inc., Crofton, MD), and Joseph A. Bucaro (Excet Inc., Springfield, VA)

Strategies for the automated detection and classification of underwater unexploded ordnance (UXO), based upon structural acoustics derived features, are currently being transitioned to autonomous underwater vehicle based sonar systems. The foundation for this transition arose, in part, from extensive laboratory investigations conducted at the Naval Research Laboratory. We discuss the evolution of structural acoustic based methodologies, including research into understanding the free-field scattering response of UXO and the coupling of these objects, under varying stages of burial, to water-saturated sediments. In addition to providing a physics-based understanding of the mechanisms contributing to the scattering response of objects positioned near the sediment–water interface, this research supports the validation of three-dimensional finite-element-based models for large-scale structural–acoustics problems. These efforts have recently culminated with the successful classification of a variety of buried UXO targets using a numerically trained relevance vector machine (RVM) classifier and the discrimination of these targets, under various burial orientations, from several objects representing both natural and manmade clutter. We conclude that this demonstration supports the transition of structural acoustic processing methodologies to maritime sonar systems for the classification of challenging UXO targets. [Work supported by ONR and SERDP.]

4:25

1pUW10. Detection and classification of marine targets buried in the sediment using structural acoustic features. Joseph Bucaro (Excet, Inc. @ Naval Res. Lab., 4555 Overlook Ave SW, Naval Res. Lab., Washington, DC 20375, joseph.bucaro.ctr@nrl.navy.mil), Brian Houston, Angie Sarkissian, Harry Simpson, Zack Waters (Naval Res. Lab., Washington, DC), Timothy Yoder (Sotera Inc. @ Naval Res. Lab., Washington, DC), and Dan Amon (Naval Res. Lab., Washington, DC)

We present research on detection and classification of underwater targets buried in a saturated sediment using structural acoustic features. These efforts involve simulations using NRL's STARS3D structural acoustics code and measurements in the NRL free-field and sediment pool facilities, off the coast of Duck, NC, and off the Coast of Panama City, FL. The measurements in the sediment pool demonstrated RVM classifiers trained using numerical data on two features—target strength correlation and elastic highlight image symmetry. Measurements off the coast of Duck were inconclusive owing to tropical storms resulting in a damaged projector. Extensive measurements were then carried out in 60 ft. of water in the Gulf using BOSS, an autonomous underwater vehicle with 40 receivers on its wings. The target field consisted of nine simulant-filled UXO and two false targets buried in the sediment and twenty proud targets. The AUV collected scattering data during north/south, east/west, and diagonal flights. We discuss the data analyzed so far from which we have extracted 3-D images and acoustic color constructs for 18 of the targets and demonstrated UXO/false target separation using a high dimensional acoustic color feature. Finally, we present related work involving targets buried in non-saturated elastic sediments. [This work is supported by ONR and SERDP.]

Contributed Papers

4:45

1pUW11. Performance metrics for depth-based signal separation using deep vertical line arrays. John K. Boyle, Gabriel P. Kniffin, and Lisa M. Zurk (Northwest Electromagnetics and Acoust. Res. Lab. (NEAR-Lab), Dept. of Elec. & Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, jboyle@pdx.edu)

A publication [McCargar & Zurk, 2013] presented a method for passive depth-separation of signals received on vertical line arrays (VLAs) deployed below the critical depth in the deep ocean. This method, based on a modified Fourier transform of the received signals from submerged targets, makes use of the depth-dependent modulation inherent in the signals due to interference between the direct and surface-reflected acoustic arrivals. Examination of the transform is necessary to determine performance of the algorithm in terms of the minimum target depth and range, array aperture, and temporal sampling. However, traditional expressions for signal sampling requirements (Nyquist sampling theorem) do not directly apply to the measured signal along a target trace due to uneven sampling in vertical angle imposed by the spatiotemporal evolution of the target track as observed on the VLA. In this paper, the effects of this uneven sampling on the ambiguity in the estimated depth (i.e., aliasing) are discussed, and expressions for the maximum snapshot length are presented and validated using simulated data

produced with a normal-mode propagation model. Initial results are presented to show the requirements for snapshot lengths and target trajectories for successful depth separation of slow moving targets at low frequencies.

5:00

1pUW12. Wideband imaging with the decomposition of time reversal operator. Chunxiao Li, Mingfei Guo, and Huancai Lu (Zhejiang Univ. of Technol., 18# ChaoWang Rd., Hangzhou, Zhejiang, Hangzhou 310014, China, chunxiaoli@zju.edu.cn)

It has been shown that the decomposition of the time reversal operator (DORT) is effective to achieve detection and selectively focusing on pointlike scatterers. Moreover, the multiplicity of the invariant of the time reversal operator for a single extended (non-pointlike) scatterer has been also revealed. In this paper, we investigate the characterization and imaging of the scatterers when an extended scatterer and a pointlike scatterer are simultaneously present. The relationship between the quality of focusing and frequency is investigated by backpropagation of singular vectors using a model of the waveguide in each frequency bin. When the extended scatterer is present, it is shown that the second singular vector can also focus on the target. However, the task of focusing can only be achieved in frequency bins with relatively large singular values. When both scatterers are simultaneously present, the singular vectors are a linear combination of the transfer vector from each scatterer. The first singular vector can achieve focusing on the extended scatterer in frequency bins with relatively large singular values. The second singular vector can approximately focus on the pointlike scatterer in frequency bins that its scattering coefficients are relatively high and the first scattering coefficient of the extended scatterer are relatively low.

Note: Payment of separate fee required to attend

MONDAY AFTERNOON, 27 OCTOBER 2014 HILBERT CIRCLE THEATER, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Musical Acoustics: Science and Performance

Uwe J. Hansen, Chair
Chemistry & Physics, Indiana State University, Terre Haute, IN 47803-2374

Invited Paper

7:00

1eID1. The physics of musical instruments with performance illustrations and a concert. Uwe J. Hansen (Dept. of Chemistry and Phys., Indiana State Univ., Indiana State Univ., Terre Haute, IN 47809, uwe.hansen@indstate.edu) and Susan Kitterman (New World Youth Orchestras, Indianapolis, IN)

Musical Instruments generally rely on the following elements for tone production: a power supply, an oscillator, a resonator, an amplifier, and a pitch control mechanism. The physical basis of these elements will be discussed for each instrument family with performance illustrations by the orchestra. Wave shapes and spectra will be shown for representative instruments. A pamphlet illustrating important elements for each instrument group will be distributed to the audience. The Science presentation with orchestral performance illustrations will be followed by a concert of the New World Youth Symphony Orchestra. This orchestra is one of three performing groups of the New World Youth Orchestras, an organization founded by Susan Kitterman in 1982. Members of the Symphony are chosen from the greater Indianapolis and Central Indiana area by audition.

Session 2aAA

Architectural Acoustics and Engineering Acoustics: Architectural Acoustics and Audio I

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

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Chair's Introduction—7:55

Invited Papers

8:00

2aAA1. Excessive reverberance in an outdoor amphitheater. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The historic Ford Theatre in Hollywood, CA, is undergoing an overall renovation and expansion. Its centerpiece is the inexplicably asymmetrical 1200 seat outdoor amphitheater, built of concrete in 1931 after the original 1920 wood structure was destroyed by a brush fire in 1929, and well before the adjacent Hollywood Freeway was nearly as noisy as now. Renovation includes reorienting seating for better symmetry while maintaining the historic concrete, and improving audio, lighting, and support spaces. Sited within an arroyo overlooking a busy highway, and in view of the Hollywood Bowl, the new design features an expanded "sound wall" that will help to mitigate highway noise while providing optimal lighting and control positions. New sound-absorptive treatments will address the Ford's excessive reverberation, currently more than might be anticipated for an entirely outdoor space. The remarkably uniform distribution of ambient noise and apparent contributions by the arroyo to the reverberation will be discussed, along with assorted design challenges.

8:20

2aAA2. Room acoustics analysis, recordings of real and simulated performances, and integration of an acoustic shell mock up with performers for evaluation of a choir shell design. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

The current renovation of the 1883 Galloway Memorial Methodist Church required the repair and replacement of a number of room finishes, as well as resolution of acoustic problems related to their choir loft. This paper will present the various approaches used to determine the best course of action using primarily an *in-situ* analysis that includes construction mockups, simulated sources, and critical listening.

8:40

2aAA3. A decade later: What we've learned from The Pritzker Pavilion at Millennium Park. Jonathan Laney, Greg Miller, Scott Pfeiffer, and Carl Giegold (Threshold Acoust., 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, jlaney@thresholdacoustics.com)

Each design and construction process yields a building and systems that respond to a particular client at a particular time. We launch these projects into the wild and all too frequently know little of their daily lives and annual cycles after that. Occasionally, though, we have the opportunity to stay close enough to watch a project wear in, weather (sometimes literally), and respond to changing client and patron dynamics over time. Such is the case with the reinforcement and enhancement systems at the Pritzker Pavilion in Chicago's Millennium Park. On a fine-grained scale, each outdoor loudspeaker is individually inspected for its condition at the end of each season. Signal-processing and amplification equipment is evaluated as well, so the overall system is maintained at a high degree of readiness and reliability. Strengths and weaknesses of these components thereby reveal themselves over time. We will discuss these technical aspects as well as changing audience behaviors, modifications made for special events, and the ways all of these factors inform the future of audio (and video) in the Park.

9:00

2aAA4. An electro-acoustic conundrum—Improving the listening experience at the Park Avenue Armory. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com) and Paul Scarbrough (Akustiks, South Norwalk, CT)

Larger than a hanger for a commercial airliner, the Park Avenue Armory occupies an entire city block in midtown Manhattan. Its massive internal volume generates reverberation time in excess of three seconds. However, it functions as a true multi-purpose venue with programming that includes dramatic performances produced by the Manchester International Festival, and musical performances sponsored by Lincoln Center. We will discuss the unique nature of the venue as well as the tools and techniques employed in staging different productions.

9:20

2aAA5. Sound reinforcement in an acoustically challenging multipurpose space. Deb Britton (K2 Audio, 4900 Pearl East Circle, Ste. 201E, Boulder, CO 80301, deb@k2audio.com)

Often times, sound system designers are dealt less than ideal cards: design a sound reinforcement system that will provide great speech intelligibility, in a highly reverberant space, without modifying any of the architectural finishes. While this can certainly be a challenge, add to those prerequisites, the additional complication of the sound system serving a multi-purpose use, where different types of presentations must take place in different locations in the space, and with varying audience sizes. This paper presents a case study of such a scenario, and describes the approach taken in order to achieve the client's goals.

9:40

2aAA6. Comparison of source stimulus input method on measured speech transmission index values of sound reinforcement systems. Neil T. Shade (Acoust. Design Collaborative, Ltd., 7509 Lhirondelle Club Rd., Ruxton, MD 21204, nts@akustx.com)

One purpose of a sound reinforcement system is to increase the talker's speech intelligibility. A common metric for speech intelligibility evaluation is the Speech Transmission Index (STI) defined by IEC-60268-16 Revision 4. The STI of a sound reinforcement system can be measured by inputting a stimulus signal into the sound system, which is modified by the system electronics, and radiated by the sound system loudspeakers to the audience seats. The stimulus signal can be input via a line level connection to the sound system or by playing the stimulus signal through a small loudspeaker that is picked-up by a sound system microphone. This latter approach factors the entire sound system signal chain from microphone input to loudspeaker output. STI measurements were performed on two sound systems, one in a reverberant room and the other in relatively non-reverberant room. Measurement results compare both signal input techniques using omnidirectional and hypercardioid sound system microphones and three loudspeakers claimed to be designed to have directivity characteristics similar to the human voice.

10:00–10:20 Break

10:20

2aAA7. Enhancements in technology for improving access to active acoustic solutions in multipurpose venues. Ronald Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com)

With advancements in digital signal processing technology and higher integration of functionality, access to active acoustics systems for multipurpose venues has been enhanced. One of the challenges with active acoustics systems in multipurpose venues is having enough control over the various acoustic environments within the same room (e.g., under balcony versus over balcony). Each area may require its own signal processing and control to be effective. Increasing the signal processing capacity to address these different environments will provide a more effective integration of the system in the room. A new signal processing platform with the flexibility to meet these needs is discussed. The new platform addresses multiple areas with concurrent processing and is integrated with a digital audio buss and a network-based control system. The system is flexible in its ability to easily expand to meet the needs of a variety of environments. Enhancing integration and flexibility of scale accelerates the potential for active systems with an attractive financial point of entry.

10:40

2aAA8. Sound levels and the risk of hearing damage at a large music college. Thomas J. Plsek (Brass, Berklee College of Music, MS 1140 Brass, 1140 Boylston St., Boston, MA 02215, tplsek@berklee.edu)

For a recent sabbatical from Berklee College of Music, my project was to study hearing loss especially among student and faculty musicians and to measure sound levels in various performance situation ranging from rehearsals to classes/labs to actual public performances. The National Institute for Occupational Safety and Health (NIOSH) recommendations (85 dBA criterion with a 3 dB exchange rate) were used to determine the daily noise dose obtained in each of the situations. In about half of the situations 100% or more of the daily noise was reached. More measuring of actual levels reached is needed as are noise dosimetry measurements over an active 12–16 hour day.

11:00

2aAA9. Development of a tunable absorber/diffuser using micro-perforated panels. Matthew S. Hildebrand (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, matt.hildebrand@wengercorp.com)

Shared rehearsal spaces are an all-too-common compromise in music education, pitting vocal, and instrumental ensembles against each other for desirable room acoustics. More than ever, adjustable acoustics are needed in music spaces. An innovative new acoustic panel system was developed with this need for flexibility in mind. Providing variable sound absorption with a truly static aesthetic,

control of reverberation time in the mid-frequency bands is ultimately handed over to the end user. New product development test methods and critical design decisions are discussed, such as curving the micro-perforated panel to improve scattering properties. *In situ* reverberation measurements are also offered against a 3D CAD model prediction using lab-tested material properties.

11:20

2aAA10. Real case measurements of inflatable membranes absorption technique. Niels W. Adelman-Larsen (Flex Acoust., Diplomvej 377, Kgs. Lyngby 2800, Denmark, nwl@flexac.com)

After some years of development of the patented technology of inflated plastic membranes for sound absorption, an actual product became available in 2012 and immediately implemented in a Danish music school. It absorbs sound somewhat linearly from 63 to 1k Hz, when active, advantageous for amplified music. The absorption coefficient is close to 0.0 when deactivated. 75.000 ft^2 of the mobile version of the innovation was employed at the Eurovision Song Contest, the second largest annual television event worldwide. This contributed to a lowering of T30 in the 63, 125, and 250 Hz octave bands from up to 13 s to below 4 s in the former-shipyard venue. The permanently installed version has been incorporated in a new theater in Korea. More detailed acoustic measurements from these cases will be presented. The technology will further be used in the new, multi-functional Dubai Opera scheduled for 2015.

11:40

2aAA11. Virtual sound images and virtual sound absorbers misinterpreted as supernatural objects. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com)

Complex sound behaviors such as echoes, reverberation, and interference patterns can be mathematically modeled using the modern concepts of virtual sound sources or virtual sound absorbers. Yet prior to the scientific wave theory of sound, these same acoustical phenomena were considered baffling, and hence led to the illusion that they were due to mysterious invisible sources. Vivid descriptions of the physical forms of echo spirits, hoofed thunder gods, and pipers' stones, as engendered from the sounds they either produced or blocked, are found in ancient myths and legends from around the world. Additional pieces of evidence attesting to these beliefs are found in archaeological remains consisting of canyon petroglyphs, cave paintings, and megalithic stone circles. Blindfolded participants in acoustic experimental set-ups demonstrated that they attributed various virtual sound effects to real sound sources and/or attenuators. Ways in which these types of sonic phenomena can be manipulated to give rise to ultra-realistic auditory illusions of actual objects even today will be discussed relative to enhancing experiences of multimedia entertainment and virtual reality. Conversely, understanding how the mind can construct psychoacoustic models inconsistent with scientific reality could serve as a lesson helping prevent the supernatural misperceptions to which our ancestors were susceptible.

TUESDAY MORNING, 28 OCTOBER 2014 LINCOLN, 8:25 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Mobile Autonomous Platforms for Bioacoustic Sensing

Holger Klinck, Cochair
Cooperative Institute for Marine Resources Studies, Oregon State University, Hatfield Marine Science Center, 2030 SE Marine Science Drive, Newport, OR 97365

David K. Mellinger, Cochair
Coop. Inst. for Marine Resources Studies, Oregon State University, 2030 SE Marine Science Dr., Newport, OR 97365

Chair's Introduction—8:25

Invited Papers

8:30

2aAB1. Real-time passive acoustic monitoring of baleen whales from autonomous platforms. Mark F. Baumgartner (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS #33, Woods Hole, MA 02543, mbaumgartner@whoi.edu)

An automated low-frequency detection and classification system (LFDCS) was developed for use with the digital acoustic monitoring (DMON) instrument to detect, classify, and report in near real time the calls of several baleen whale species, including fin, humpback, sei, bowhead, and North Atlantic right whales. The DMON/LFDCS has been integrated into the Slocum glider and APEX profiling float, and integration projects are currently underway for the Liquid Robotics wave glider and a moored buoy. In a recent

evaluation study, two gliders reported over 25,000 acoustic detections attributed to fin, humpback, sei, and North Atlantic right whales over a 3-week period during late fall in the Gulf of Maine. The overall false detection rate for individual calls was 14%, and for right, humpback, and fin whales, false predictions of occurrence during 15-minute reporting periods were 5% or less. Agreement between acoustic detections and visual sightings from concurrent aerial and shipboard surveys was excellent (9 of 10 visual detections were accompanied by real-time acoustic detections of the same species by a nearby glider). We envision that this autonomous acoustic monitoring system will be a useful tool for both marine mammal research and mitigation applications.

8:50

2aAB2. Detection, bearing estimation, and telemetry of North Atlantic right whale vocalizations using a wave glider autonomous vehicle. Harold A. Cheyne (Lab of Ornithology, Cornell Univ., 95 Brown Rd., Rm. 201, Ithaca, NY 14850, haroldcheyne@gmail.com), Charles R. Key, and Michael J. Satter (Leidos, Long Beach, MS)

Assessing and mitigating the effects of anthropogenic noise on marine mammals is limited by the typically employed technologies of archival underwater acoustic recorders and towed hydrophone arrays. Data from archival recorders are analyzed months after the activity of interest, so assessment occurs long after the events and mitigation of those activities is impossible. Towed hydrophone arrays suffer from nearby ship and seismic air gun noise, and they require substantial on-board human and computing resources. This work has developed an acoustic data acquisition, processing, and transmission system for use on a Wave Glider, to overcome these limitations by providing near real-time marine mammal acoustic data from a portable and persistent autonomous platform. Sea tests have demonstrated the proof-of-concept with the system recording four channels of acoustic data and transmitting portions of those data via satellite. The system integrates a detection-classification algorithm on-board, and a beam-forming algorithm in the shore-side user interface, to provide a user with aural and visual review tools for the detected sounds. Results from a two-week deployment in Cape Cod Bay will be presented and future development directions will be discussed.

9:10

2aAB3. Shelf-scale mapping of fish sound production with ocean gliders. David Mann (Loggerhead Instruments Inc., 6576 Palmer Park Circle, Sarasota, FL 34238, dmann@loggerhead.com), Carrie Wall (Univ. of Colorado at Boulder, Boulder, CO), Chad Lembke, Michael Lindemuth (College of Marine Sci., Univ. of South Florida, St.. Petersburg, FL), Ruoying He (Dept Marine, Earth, and Atmospheric Sci., NC State Univ., Raleigh, NC), Chris Taylor, and Todd Kellison (Beaufort Lab., NOAA Fisheries, Beaufort, NC)

Ocean gliders are a powerful platform for collecting large-scale data on the distribution of sound-producing animals while also collecting environmental data that may influence their distribution. Since 2009, we have performed extensive mapping on the West Florida Shelf with ocean gliders equipped with passive acoustic recorders. These missions have revealed the distribution of red grouper as well as identified several unknown sounds likely produced by fishes. In March 2014, we ran a mission along the shelf edge from Cape Canaveral, FL to North Carolina to map fish sound production. The Gulf Stream and its strong currents necessitated a team effort with ocean modeling to guide the glider successfully to two marine protected areas. This mission also revealed large distributions of unknown sounds, especially on the shallower portions of the shelf. Gliders provide valuable spatial coverage, but because they are moving and most fish have strong diurnal sound production patterns, data analysis on presence and absence must be made carefully. In many of these cases, it is best to use a combination of platforms, including fixed recorders and ocean profilers to measure temporal patterns of sound production.

9:30

2aAB4. The use of passively drifting acoustic recorders for bioacoustic sensing. Jay Barlow, Emily Griffiths, and Shannon Rankin (Marine Mammal and Turtle Div., NOAA-SWFSC, 8901 La Jolla Shores Dr., La Jolla, CA 92037, jay.barlow@noaa.gov)

Passively drifting recording systems offer several advantages over autonomous underwater or surface vessels for mobile bioacoustic sensing in the sea. Because they lack of any propulsion, self noise is minimized. Also, vertical hydrophone arrays are easy to implement, which is useful in estimating the distance to specific sound sources. We have developed an inexpensive (<$5000) Drifting Acoustic Spar Buoy Recorder (DASBR) that features up to 1 TB of stereo recording capacity and a bandwidth of 10 Hz–96 kHz. Given their low cost, many more recorders can be deployed to achieve greater coverage. The audio and GPS recording system floats at the surface, and the two hydrophones (at 100 m) are de-coupled from wave action by a dampner disk and an elastic cord. During a test deployment in the Catalina Basin (Nov 2013) we collected approximately 1200 hours of recordings using 5 DASBRs recording at 192 kHz sampling rate. Each recorder was recovered (using GPS and VHF locators) and re-deployed 3–4 times. Dolphin whistles and echo-location clicks were detectable approximately half of the total recording time. Cuvier's beaked whales were also detected on three occasions. Cetacean density estimation and ocean noise measurements are just two of many potential uses for free-drifting recorders.

9:50

2aAB5. Small cetacean monitoring from surface and underwater autonomous vehicles. Douglas M. Gillespie, Mark Johnson (Sea Mammal Res. Unit, Univ. of St. Andrews, Gatty Marine Lab., St Andrews, Fife KY16 8LB, United Kingdom, dg50@st-andrews.ac.uk), Danielle Harris (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, Fife, United Kingdom), and Kalliopi Gkikopoulou (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, United Kingdom)

We present results of Passive Acoustic surveys conducted from three types of autonomous marine vehicles, two submarine gliders and a surface wave powered vehicle. Submarine vehicles have the advantage of operating at depth, which has the potential to increase detection rate for some species. However, surface vehicles equipped with solar panels have the capacity to carry a greater payload and currently allow for more on board processing which is of particular importance for high frequency odontocete species. Surface vehicles are also more suited to operation in shallow or coastal waters. We describe the hardware and software packages developed for each vehicle type and give examples of the types of data retrieved both through real time telemetry and recovered post deployment. High frequency echolocation clicks and whistles have been successfully detected from all vehicles. Noise levels varied considerably between vehicle types, though all were subject to a degree of mechanical noise from the vehicle itself.

10:10–10:35 Break

10:35

2aAB6. A commercially available sound acquisition and processing board for autonomous passive acoustic monitoring platforms.
Haru Matsumoto, Holger Klinck, David K. Mellinger (CIMRS, Oregon State Univ., 2115 SE OSU Dr., Newport, OR 97365, haru.matsumoto@oregonstate.edu), and Chris Jones (Embedded Ocean Systems, Seattle, WA)

The U.S. Navy is required to monitor marine mammal populations in U.S. waters to comply with regulations issued by federal agencies. Oregon State University and Embedded Ocean Systems (EOS) co-developed a passive acoustic data acquisition and processing board called Wideband Intelligent Signal Processor and Recorder (WISPR). This low-power, small-footprint system is suitable for autonomous platforms with limited battery and space capacity, including underwater gliders and profiler floats. It includes a high-performance digital signal processor (DSP) running the uClinux operating system, providing extensive flexibility for users to configure or re-program the system's operation. With multiple WISPR-equipped mobile platforms strategically deployed in an area of interest, operators on land or at sea can now receive information in near-real time about the presence of protected species in the survey area. In April 2014, WISPR became commercially available via EOS. We are implementing WISPR in the Seaglider and will conduct a first evaluation test off the coast of Oregon in September. System performance, including system noise interference, flow noise, power consumption, and file compression rates in the data-logging system, will be discussed. [Funding from the US Navy's Living Marine Resources Program.]

10:55

2aAB7. Glider-based passive acoustic marine mammal detection. John Hildebrand, Gerald L. D'Spain, and Sean M. Wiggins (Scripps Inst. of Oceanogr., UCSD, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu)

Passive acoustic detection of delphinid sounds using the Wave Glider (WG) autonomous near-surface vehicle was compared with a fixed bottom-mounted autonomous broadband system, the High-frequency Acoustic Recording Package (HARP). A group of whistling and clicking delphinids was tracked using an array of bottom-mounted HARPs, providing ground-truth for detections from the WG. Whistles in the 5–20 kHz band were readily detected by the bottom HARPs as the delphinids approached, but the WG revealed only a brief period with intense detections as the animals approached within ~500 m. Refraction due to acoustic propagation in the thermocline provides an explanation for why the WG may only detect whistling delphinids at close range relative to the long-range detection capabilities of the bottom-mounted HARPs. This work demonstrated that sound speed structure plays an important role in determining detection range for high-frequency-calling marine mammals by autonomous gliders and bottom-mounted sensors.

Contributed Papers

11:15

2aAB8. Acoustic seagliders for monitoring marine mammal populations. Lora J. Van Uffelen (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96815, loravu@hawaii.edu), Erin Oleson (Cetacean Res. Program, NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), Bruce Howe, and Ethan Roth (Ocean and Resources Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

A DMON digital acoustic monitoring device has been integrated into a Seaglider with the goal of passive, persistent acoustic monitoring of cetacean populations. The system makes acoustic recordings as it travels in a sawtooth pattern between the surface and up to 1000 m depth. It includes three hydrophones, located in the center of the instrument and on each wing. An onboard real-time detector has been implemented to record continuously after ambient noise has risen above a signal-to-noise (SNR) threshold level, and the glider transmits power spectra of recorded data back to a shore-station computer via iridium satellite after each dive. The glider pilot has the opportunity to set parameters that govern the amount of data recorded, thus managing data storage and therefore the length of a mission. This system was deployed in the vicinity of the Hawaiian Islands to detect marine mammals as an alternative or complement to conventional ship-based survey methods. System design and implementation will be described and preliminary results will be presented.

11:30

2aAB9. Prototype of a linear array on an autonomous surface vehicle for the register of dolphin displacement patterns within a shallow bay.
Eduardo Romero-Vivas, Fernando D. Von Borstel-Luna (CIBNOR, Instituto Politécnico Nacional 195, Playa Palo de Santa Rita Sur, La Paz, BCS 23090, Mexico, evivas@cibnor.mx), Omar A. Bustamante, Sergio Beristain (Acoust. Lab, ESIME, IPN, IMA, Mexico City, Mexico), Miguel A. Porta-Gándara, Francisco Villa Médina, and Joaquín Gutiérrez-Jagüey (CIBNOR, La Paz, BCS, Mexico)

A semi-resident population of tursiops has been reported in the south of La Paz bay in Baja California Sur, Mexico, where specific zones for social, feeding and resting behaviors have been detected. Nevertheless, increasing human activities and new constructions are attributed to have shifted the areas of their main activity. Therefore, it becomes important to study displacement patterns of dolphins within the bay and their spatial relationship to maritime traffic and other sources of anthropogenic noise. A prototype of an Autonomous Surface Vehicle (ASV) designed for shallow water bathymetry has been adapted to carry a linear array of hydrophones previously reported for the localization of dolphins from their whistles. Conventional beam-forming algorithms and electrical steering are used to find Direction of Arrival (DOA) of the sound sources. The left-right ambiguity typical of a linear array and front-back lobes for sound sources located at end-fire can be resolved by the trajectory of the ASV. Geo-referenced positions and bearing of the array, provided by the Inertial Measurement Unit of the ASV, along with DOA for various positions allows triangulating and mapping the sound sources. Results from both, controlled experiments using geo-referenced know sources, and field trials within the bay, are presented.

11:45

2aAB10. High-frequency observations from mobile autonomous platforms. Holger Klinck, Haru Matsumoto, Selene Fregosi, and David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, Holger.Klinck@oregonstate.edu)

With increased human use of US coastal waters—including use by renewable energy activities such as the deployment and operation of wind, wave, and tidal energy converters—the issue of potential negative impacts on coastal ecosystems arises. Monitoring these areas efficiently for marine mammals is challenging. Recreational and commercial activities (e.g., fishing) can hinder long-term operation of fixed moored instruments. Additionally these shallow waters are often utilized by high-frequency cetaceans (e.g., harbor porpoises) which can only be acoustically detected over short distances of a few hundred meters. Mobile acoustic platforms are a useful tool to survey these areas of concern with increased temporal and spatial resolution compared to fixed systems and towed arrays. A commercially available acoustic recorder (type Song Meter SM2+, Wildlife Acoustics, Inc.) featuring sampling rates up to 384 kHz was modified and implemented on an autonomous underwater vehicle (AUV) as well as an unmanned surface vehicle (USV) and tested in the field. Preliminary results indicate that these systems are effective at detecting the presence of high-frequency cetaceans such as harbor porpoises. Potential applications, limitations, and future directions of this technology will be discussed. [Project partly supported by ONR and NOAA.]

TUESDAY MORNING, 28 OCTOBER 2014 INDIANA G, 8:25 A.M. TO 12:00 NOON

Session 2aAO

Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Parameter Estimation in Environments That Include Out-of-Plane Propagation Effects

Megan S. Ballard, Cochair
Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78758

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

Chair's Introduction—8:25

Invited Papers

8:30

2aAO1. Estimating waveguide parameters using horizontal and vertical arrays in the vicinity of horizontal Lloyd's mirror in shallow water. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

When shallow water internal waves approach a source-receiver track, the interference between the direct and horizontally refracted acoustic paths from a broadband acoustic source was previously shown to form Horizontal Lloyd's mirror (Badiey *et al*. J. Acoust. Soc. Am. 128(4), EL141–EL147, 2011). While the modal interference structure in the vertical plane may reveal arrival time for the out of plane refracted acoustic wave front, analysis of moving interference pattern along the horizontal array allows measurement of the angle of horizontal refraction and the speed of the nonlinear internal wave (NIW) in the horizontal plane. In this paper we present a full account of the movement of NIW towards a source-receive track and how we can use the received acoustic signal on an L-shaped array to estimate basic parameters of the waveguide and obtain related temporal and spatial coherence functions particularly in the vicinity of the formation of the horizontal Lloyd mirror. Numerical results using Vertical Modes and Horizontal Rays as well as 3D PE calculations are carried out to explain the experimental observations. [Work supported by ONR 322OA.]

8:50

2aAO2. Slope inversion in a single-receiver context for three-dimensional wedge-like environments. Frédéric Sturm (LMFA (UMR 5509 ECL-UCBL1-INSA de Lyon), Ecole Centrale de Lyon, Ctr. Acoustique, Ecole Centrale de Lyon, 36, Ave. Guy de Collongue, Ecully 69134, France, frederic.sturm@ec-lyon.fr) and Julien Bonnel (Lab-STICC (UMR CNRS 6285), ENSTA Bretagne, Brest Cedex 09, France)

In a single-receiver context, time-frequency (TF) analysis can be used to analyze modal dispersion of low-frequency broadband sound pulses in shallow-water oceanic environments. In a previous work, TF analysis was used to study the propagation of low-frequency broadband pulses in three-dimensional (3-D) shallow-water wedge waveguides. Of particular interest is that TF analysis turns out to be a suitable tool to better understand, illustrate and visualize 3-D propagation effects for such wedge-like environments. In the present work, it is shown that TF analysis can also be used at the core of an inversion scheme to estimate the slope of the seabed in a same single hydrophone receiving configuration and for similar 3-D wedge-shaped waveguides. The inversion algorithm proposed, based on a masking process, focuses on specific parts of the TF domain where modal energy is concentrated. The criterion used to quantify the match between the received signal and the replicas by a fully 3-D parabolic equation code, is defined as the amount of measured time-frequency energy integrated inside the masks. Its maximization is obtained using an exhaustive search. The method is first benchmarked on numerical simulations and then successfully applied on experimental small-scale data.

9:10

2aAO3. Effects of environmental uncertainty on source range estimates from horizontal multipath. Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

A method has been developed to estimate source range in continental shelf environments that exhibit three-dimensional propagation effects [M. S. Ballard, J. Acoust. Soc. Am. 134, EL340–EL343, 2013]. The technique exploits measurements recorded on a horizontal line array of a direct path arrival, which results from sound propagating across the shelf to the receiver array, and a refracted path arrival, which results from sound propagating obliquely upslope and refracting back downslope to the receiver array. A hybrid modeling approach using vertical modes and horizontal rays provides the ranging estimate. According to this approach, rays are traced in the horizontal plane with refraction determined by the modal phase speed. Invoking reciprocity, the rays originate from the center of the array and have launch angles equal to the estimated bearing angles of the direct and refracted paths. The location of the source in the horizontal plane is estimated from the point where the rays intersect. In this talk, the effects of unknown environmental parameters, including the sediment properties and the water-column sound-speed profile, on the source range estimate are discussed. Error resulting from uncertainty in the measured bathymetry and location of the receiver array will also be addressed. [Work supported by ONR.]

Contributed Papers

9:30

2aAO4. Acoustical observation of the estuarine salt wedge at low-to-mid-frequencies. D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, 73 Hanapepe Loop, Honolulu, HI 96825, dbreeder@nps.edu)

The estuarine environment often hosts a salt wedge, the stratification of which is a function of the tide's range and speed of advance, river discharge volumetric flow rate and river mouth morphology. Competing effects of temperature and salinity on sound speed control the degree of acoustic refraction occurring along an acoustic path. A field experiment was carried out in the Columbia River to test the hypothesis that the estuarine salt wedge is acoustically observable in terms of low-to-mid-frequency acoustic propagation. Linear frequency-modulated (LFM) acoustic signals in the 500–2000 Hz band were collected during the advance and retreat of the salt wedge during May 27–28, 2013. Results demonstrate that the three-dimensional salt wedge front is the dominant physical feature controlling acoustic propagation in this environment: received signal energy is relatively stable under single-medium conditions before and after the passage of the salt wedge front, but suffers a 10–15 dB loss as well as increased variance during salt wedge front passage due to 3D refraction and scattering. Physical parameters (i.e., temperature, salinity, current, and turbulence) and acoustic propagation modeling corroborate and inform the acoustic observations.

9:45

2aAO5. A hybrid approach for estimating range-dependent properties of shallow water environments. Michael Taroudakis and Costas Smaragdakis (Mathematics and Appl. Mathematics & IACM, Univ. of Crete and FORTH, Knossou Ave., Heraklion 71409, Greece, taroud@math.uoc.gr)

H hybrid approach based on statistical signal characterization and a linear inversion scheme for the estimation of range dependent sound speed profiles of compact support in shallow water is presented. The approach is appropriate for ocean acoustic tomography when there is a single receiver available, as the first stage of the method is based on the statistical characterization of a single reception using wavelet transform to associate the signal with a set of parameters describing the statistical features of its wavelet sub-band coefficients. A non-linear optimization algorithm is then applied to associate these features with range-dependent sound speed profile in the water column. This inversion method is restricted to cases where the range dependency is of compact support. At the second stage a linear inversion scheme based on modal arrivals identification and a first order perturbation formula to associate sound speed differences with modal travel time perturbations is applied to fine tune the results obtained by the optimization scheme. A second restriction of this stage is that mode identification is necessary. If this assumption is fulfilled the whole scheme may be applied in ocean acoustic tomography for the retrieval of three-dimensional features, combining inversion results at various slices.

10:00–10:15 Break

Invited Papers

10:15

2aAO6. Three-dimensional acoustics in basin scale propagation. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com) and Richard L. Campbell (OASIS Inc., Seattle, U.S. Minor Outlying Islands)

Long-range, basin-scale acoustic propagation has long been considered deep water and well represented by the two-dimensional numerical solution (range/depth) of wave equation. Ocean acoustic tomography has even recently been demonstrated to be insensitive to the three-dimensional affects of refraction and diffraction (Dushaw, JASA 2014). For frequencies below 50 Hz, where volume attenuation is negligible, the approximation that all propagation of significance is in the plane begins to break down. When examining very long-range propagation in situations where the source/receiver are not specifically selected for open water paths, 3D effects can dominate. Seamounts and bathymetry rises cause both refraction away from the shallowing seafloor and diffraction behind sharp edges. In this paper a set of recent observations, many from the International Monitoring System (IMS) of the United Nations Comprehensive Test Ban Treaty Organization (CTBTO) will be presented, demonstrating observations that are not well explained by Nx2D acoustic propagation. The Peregrine PE model, a recent recoding of RAM in C, has been extended to include 3D split-step Pade propagation and will be used to demonstrate how 3D acoustic propagation affects help explains some of the observations.

10:35

2aAO7. Sensitivity analysis of three-dimensional sound pressure fields in complex underwater environments. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

A sensitivity kernel for sound pressure variability due to variations of index of refraction is derived from a higher-order three-dimensional (3D) split-step parabolic-equation (PE) solution of the Helmholtz equation. In this study, the kernel is used to compute the acoustic sensitivity field between a source and a receiver in a 3D underwater environment, and to quantify how much of the medium change can cause significant consequence on received acoustic signals. Using the chain rule, the dynamics of sensitivity fields can be connected to the dynamics of ocean processes. This talk will present numerical examples of sound propagation in submarine canyons and continental slopes, where the ocean dynamics cause strong spatial and temporal variability in sound pressure. Using the sensitivity kernel technique, we can analyze the spatial distribution and the temporal evolution of the acoustic sensitivity fields in these geologically and topographically complex environments. The paper will also discuss other applications of this sound pressure sensitivity kernel, including uncertainty quantification of transmission loss prediction and adjoint models for 3D acoustic inversions. [Work supported by the ONR.]

10:55

2aAO8. Sensitivity analysis of the image source method to out-of-plane effects. Samuel Pinson (Laboratório de Vibrações e Acústica, Universidade Federal de Santa Catarina, LVA Dept de Engenharia Mecânica, UFSC, Bairro Trindade, Florianópolis, SC 88040-900, Brazil, samuelpinson@yahoo.fr) and Charles W. Holland (Penn State Univ., State College, PA)

In the context of seafloor characterization, the image source method is a technique to estimate the sediment sound-speed profile from broadband seafloor reflection data. Recently the method has been extended to treat non-parallel layering of the sediment stack. In using the method with measured data, the estimated sound-speed profiles are observed to exhibit fluctuations. These fluctuations may be partially due to violation of several assumptions: (1) the layer interfaces are smooth with respect to the wavelength and (2) out-of-plane effects are negligible. In order to better understand the impact these effects, the sensitivity of the image source method to roughness and out-of-plane effects are examined.

Contributed Papers

11:15

2aAO9. Results of matched-field inversion in a three-dimensional oceanic environment ignoring horizontal refraction. Frédéric Sturm (LMFA (UMR 5509 ECL-UCBL1-INSA de Lyon), Ecole Centrale de Lyon, Ctr. Acoustique, Ecole Centrale de Lyon, 36, Ave. Guy de Collongue, Ecully 69134, France, frederic.sturm@ec-lyon.fr) and Alexios Korakas (Lab-STICC (UMR6285), ENSTA Bretagne, Brest Cedex 09, France)

For some practical reasons, inverse problems in ocean acoustics are often based on 2-D modeling of sound propagation, hence ignoring 3-D propagation effects. However, the acoustic propagation in shallow-water environments, such as the continental shelf, may be strongly affected by 3-D effects, thus requiring 3-D modeling to be accounted for. In the present talk, the feasibility and the limits of an inversion in fully 3-D oceanic environments assuming 2-D propagation are investigated. A simple matched-field inversion procedure implemented in a Bayesian framework and based on the exhaustive search of the parameter space is used. The study is first carried out on a well-established wedge-like synthetic test case, which exhibits well-known 3-D effects. Both synthetic data and replica are generated using a parabolic-equation-based code. This approach highlights the relevance of using 2-D propagation models when inversions are performed at relatively short ranges from the source. On the other hand, important mismatch occurs when inverting at farther ranges, demonstrating that the use of fully 3-D forward models is required. Results of inversion on experimental small-scale data, based on a subspace approach as suggested by the preliminary study made on the synthetic test case, are presented.

11:30

2aAO10. Measurements of sea surface effects on the low-frequency acoustic propagation in shallow water. Altan Turgut, Marshall H. Orr (Acoust. Div., Naval Res. Lab, Acoust. Div., Code 7161, Washington, DC 20375, altan.turgut@nrl.navy.mil), and Jennifer L. Wylie (Fellowships Office, National Res. Council, Washington, DC)

In shallow water, spatial and temporal variability of the water column often restricts accurate estimations of bottom properties from low-frequency acoustic data, especially under highly active oceanographic conditions during the summer. These effects are reduced under winter conditions having a more uniform sound-speed profile. However, during the RAGS03 winter experiment, significant low-frequency (200–500 Hz) acoustic signal degradations have been observed on the New Jersey Shelf, especially in the presence of frequently occurring winter storms. Both in-plane and out-of-plane propagation effects were observed on three moored VLAs and one bottom-moored HLA. These effects were further analyzed using 3-D PE simulations with inputs from a 3-D time-evolving surface gravity wave model. It is shown that higher-order acoustic modes are highly scattered at high sea states and out-of-plane propagation effects become important when surface-wave fronts are parallel to the acoustic propagation track. In addition, 3-D propagation effects on the source localization and geoacoustic inversions are investigated using the VLA data with/without the presence of winter storms. [Work supported by ONR.]

11:45

2aAO11. Effects of sea surface roughness on the mid-frequency acoustic pulse decay in shallow water. Jennifer Wylie (National Res. Council, 6141 Edsall Rd., Apt. H, Alexandira, VA 22304, jennie.wylie@gmail.com) and Altan Turgut (Acoust. Div., Naval Res. Lab., Washington, DC)

Recent and ongoing efforts to characterize sea bed parameters from measured acoustic pulse decay have neglected the effects of sea surface roughness. In this paper, these effects are investigated using a rough surface version of RAMPE, RAMSURF, and random rough surface realizations, calculated from a 2D JONSWAP sea surface spectrum with directional spreading. Azimuthal dependence is investigated for sandy bottoms and found that the rate of pulse decay increases when the surface wave fronts are perpendicular to the path of acoustic propagation and higher significant wave height results in higher decay rates. Additionally, the effects from sea surface roughness are found to vary with different waveguide parameters including but not limited to sound speed profile, water depth, and seabed properties. Of particular interest are the combined effects of sea bed properties and rough sea surfaces. It is shown that when clay like sediments are present, higher-order modes are strongly attenuated and effects due to interaction with the rough sea surface are less pronounced. Finally, possible influences of sea-state and 3D out-of-plane propagation effects on the seabed characterization efforts will be discussed. [Work supported by ONR.]

TUESDAY MORNING, 28 OCTOBER 2014 INDIANA A/B, 7:55 A.M. TO 12:10 P.M.

Session 2aBA

Biomedical Acoustics: Quantitative Ultrasound I

Michael Oelze, Cochair
UIUC, 405 N. Mathews, Urbana, IL 61801

Jonathan Mamou, Cochair
F. L. Lizzi Center for Biomedical Engineering, Riverside Research, 156 William St., 9th Floor, New York, NY 10038

Chair's Introduction—7:55

Invited Papers

8:00

2aBA1. Myocardial tissue characterization: Myofiber-induced ultrasonic anisotropy. James G. Miller (Phys., Washington U Saint Louis, Box 1105, 1 Brookings Dr., Saint Louis, MO 63130, james.g.miller@wustl.edu) and Mark R. Holland (Radiology and Imaging Sci., Indiana Univ. School of Medicine, Indianapolis, IN)

One goal of this invited presentation is illustrate the capabilities of quantitative ultrasonic imaging (tissue characterization) to determine local myofiber orientation using techniques applicable to clinical echocardiographic imaging. Investigations carried out in our laboratory in the late 1970s were perhaps the first reported studies of the impact on the ultrasonic attenuation of the angle between the incoming ultrasonic beam and the local myofiber orientation. In subsequent studies, we were able to show that the ultrasonic backscatter exhibits a maximum and the ultrasonic attenuation exhibits a minimum when the sound beam is perpendicular to myofibers, whereas the attenuation is maximum and the backscatter is minimum for parallel insonification. Results from our laboratory demonstrate three broad areas of potential contribution derived from quantitative ultrasonic imaging and tissue characterization: (1) improved diagnosis and patient management, such as monitoring alterations in regional myofiber alignment (for example, potentially in diseases such as hypertrophic cardiomyopathy), (2) improved echocardiographic imaging, such as reduced lateral wall dropout in short axis echocardiographic images, and (3) improved understanding of myocardial physiology, such as contributing to a better understanding of myocardial twist resulting from the layer-dependent helical configuration of cardiac myofibers. [NIH R21 HL106417.]

8:20

2aBA2. Quantitative ultrasound for diagnosing breast masses considering both diffuse and non-diffuse scatterers. James Zagzebski, Ivan Rosado-Mendez, Haidy Gerges-Naisef, and Timothy Hall (Medical Phys., Univ. of Wisconsin, 1111 Highland Ave., Rm. L1 1005, Madison, WI 53705, jazagzeb@wisc.edu)

Quantitative ultrasound augments conventional ultrasound information by providing parameters derived from scattering and attenuation properties of tissue. This presentation describes our work estimating attenuation (ATT) and backscatter coefficients (BSC), and computing effective scatterer sizes (ESD) to differentiate benign from malignant breast masses. Radio-frequency echo data are obtained from patients scheduled for biopsy of suspicious masses following an institutional IRB approved protocol. A Siemens S2000 equipped with a linear array and recently a volume scanner transducer is employed. Echo signal power spectra are computed from the tissue and from the same depth in a reference phantom having accurately measured acoustic properties. Ratios of the tissue-to-reference power spectra enable tissue ATT and BSC's to be estimated. ESD's are then computed by fitting BSC vs. frequency results to a size-dependent scattering model. A heterogeneity index HDI expresses variability of the ESD over the tumor area. In preliminary data from 35 patients, a Bayesian classifier incorporating ATT, ESD, and HDI successfully differentiated malignant masses from fibroadenomas. Future work focuses on analysis methods when diffuse scattering and stationary signal conditions, implicitly assumed in the power spectra calculations, are not present. This approach tests for signal coherence and generates new parameters that characterize these scattering conditions.

8:40

2aBA3. Quantitative ultrasound translates to human conditions. William O'Brien (Elec. and Comput. Eng., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, wdo@uiuc.edu)

Two QUS studies will be discussed that demonstrate significant potential for translation to human conditions. One of the studies deals with the early detection of spontaneous preterm birth (SPTB). In a cohort of 68 adult African American women, each agreed to undergo up to five transvaginal ultrasound examinations for cervical ultrasonic attenuation (at 5 MHz) and cervical length between 20 and 36 weeks gestation (GA). At 21 weeks GA, the women who delivered preterm had a lower mean attenuation (1.02 ± 0.16 dB/cm MHz) than the women delivering at term (1.39 ± 0.095 dB/cm MHz), $p = 0.041$. Cervical length at 21 weeks was not significantly different between groups. Attenuation risk of SPTB (1.2 dB/cm MHz threshold at 21 weeks): specificity = 83.3%, sensitivity = 65.4%. The other QUS study deals with the early detection of nonalcoholic fatty liver disease (NAFLD). Liver attenuation (ATN) and backscattered coefficients (BSC) were assessed at 3 MHz and compared to the liver MR-derived fat fraction (FF) in a cohort of 106 adult subjects. At a 5% FF (for NAFLD, FF $\geq$ 5%), an ATN threshold of 0.78 dB/cm MHz provided a sensitivity of 89%, and specificity of 84%, whereas a BSC threshold of 0.0028/cm-sr provided a sensitivity of 92% and specificity of 96%.

9:00

2aBA4. Quantitative-ultrasound detection of cancer in human lymph nodes based on support vector machines. Jonathan Mamou, Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@rri-usa.org), Alain Coron (Laboratoire d'Imagerie Biomédicale, Sorbonne Universités and UPMC Univ Paris 06 and CNRS and INSERM, Paris, France), Emi Saegusa-Beecroft (Dept. of Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), Thanh Minh Bui (Laboratoire d'Imagerie Biomédicale, Sorbonne Universités and UPMC Univ Paris 06 and CNRS and INSERM, Paris, France), Michael L. Oelze (BioAcoust. Res. Lab., Univ. of Illinois, Urbana-Champaign, IL), Eugene Yanagihara (Dept. of Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), Lori Bridal (Laboratoire d'Imagerie Biomédicale, Sorbonne Universités and UPMC Univ Paris 06 and CNRS and INSERM, Paris, France), Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan), Junji Machi (Dept. of Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), and Ernest J. Feleppa (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY)

Histological assessment of lymph nodes excised from cancer patients suffers from an unsatisfactory rate of false-negative determinations. We are evaluating high-frequency quantitative ultrasound (QUS) to detect metastatic regions in lymph nodes freshly excised from cancer patients. Three-dimensional (3D) RF data were acquired from 289 lymph nodes of 82 colorectal-, 15 gastric-, and 70 breast-cancer patients with a custom scanner using a 26-MHz, single-element transducer. Following data acquisition, individual nodes underwent step-sectioning at 50-μm to assure that no clinically significant cancer foci were missed. RF datasets were analyzed using 3D regions-of-interest that were processed to yield 13 QUS estimates including spectral-based and envelope-statistics-based parameters. QUS estimates are associated with tissue microstructure and are hypothesized to provide contrast between non-cancerous and cancerous regions. Leave-one-out classifications, ROC curves, and areas under the ROC (AUC) were used to compare the performance of support vector machines (SVMs) and step-wise linear discriminant analyses (LDA). Results showed that SVM performance (AUC = 0.87) was superior to LDA performance (AUC = 0.78). These results suggest that QUS methods may provide an effective tool to guide pathologists towards suspicious regions and also indicate that classification accuracy can be improved using sophisticated and robust classification tools. [Supported in part by NIH grant CA100183.]

9:20

2aBA5. Quantitative ultrasound assessment of tumor responses to chemotherapy using a time-integrated multi-parameter approach. Hadi Tadayyon, Ali Sadeghi-Naini, Lakshmanan Sannachi, and Gregory Czarnota (Dept. of Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, gregory.czarnota@sunnybrook.ca)

Radiofrequency ultrasound data were collected from 60 breast cancer patients prior to treatment and at during the onset of their several-month treatment, using a clinical ultrasound scanner operating a ~7 MHz linear array probe. ACE, SAS, spectral, and BSC parameters were computed from 2×2 mm RF segments within the tumor region of interest (ROI) and averaged over all segments to obtain a mean value for the ROI. The results were separated into two groups—responders and non-responders—based on the ultimate clinical/pathologic response based on residual tumor size and tumor cellularity. Using a single parameter approach, the best prediction of response was achieved using the ACE parameter (76% accuracy at week 1). In general, more favorable classifications were achieved using spectral parameter combinations (82% accuracy at week 8), compared to BSC parameter combinations (73% accuracy). Using the multi-parameter approach, the best prediction was achieved using the set [MBF, SS, SAS, ACE] and by combining week 1 QUS data with week 4 QUS data to predict the response at week 4, providing accuracy as high as 91%. The proposed QUS method may potentially provide early response information and guide cancer therapies on an individual patient basis.

9:40

2aBA6. Quantitative ultrasound methods for uterine-cervical assessment. Timothy J. Hall (Medical Phys., Univ. of Wisconsin, 1005 WIMR, 1111 Highland Ave., Madison, WI 53705, tjhall@wisc.edu), Helen Feltovich (Medical Phys., Univ. of Wisconsin, Park City, Utah), Lindsey C. Carlson, Quinton Guerrero, Ivan M. Rosado-Mendez, and Bin Huang (Medical Phys., Univ. of Wisconsin, Madison, WI)

The cervix is a remarkable organ. One of its tasks is to remain firm and "closed" (5 mm diameter cervical canal) prior to pregnancy. Shortly after conception the cervix begins to soften through collagen remodeling and increased hydration. As the fetus reaches full-term there is a profound breakdown in the collagen structure. At the end of this process, the cervix is as soft as warm butter and the cervical canal has dilated to about 100 mm diameter. Errors in timing of this process are a cause for preterm birth, which has a cascade of life-threatening consequences. Quantitative ultrasound is well-suited to monitoring these changes. We have demonstrated the ability to accurately assess the elastic properties and acoustic scattering properties (anisotropy in backscatter and attenuation) of the cervix in non-pregnant hysterectomy specimens and in third trimester pregnancy. We've shown that acoustic and mechanical properties vary along the length of the cervix. When anisotropy and spatially variability are accounted for, there are clear differences in parameter values with subtle differences in softening. We are corroborating acoustic observations with nonlinear optical microscopy imaging for a reality check on underlying tissue structure. This presentation will provide an overview of this effort.

10:00–10:10 Break

10:10

2aBA7. Characterization of anisotropic media with shear waves. Matthew W. Urban, Sara Aristizabal, Bo Qiang, Carolina Amador (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), John C. Brigham (Dept. of Civil and Environ. Eng., Dept. of BioEng., Univ. of Pittsburgh, Pittsburgh, PA), Randall R. Kinnick, Xiaoming Zhang, and James F. Greenleaf (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

In conventional shear wave elastography materials are assumed to be linear, elastic, homogeneous, and isotropic. These assumptions are important to account for in certain tissues because they are not always appropriate. Many tissues such as skeletal muscle, the kidney, and the myocardium are anisotropic. Shear waves can be used to investigate the directionally dependent mechanical properties of anisotropic media. To study these tissues in a systematic way and to account for the effects of the anisotropic architecture, laboratory-based phantoms are desirable. We will report on several phantom-based approaches for studying shear wave anisotropy, assuming that these materials are transversely isotropic. Phantoms with embedded fibers were used to mimic anisotropic tissues. Homogeneous phantoms were compressed to induce transverse isotropy according to the acoustoelastic phenomenon, which is related to nonlinear behavior of the materials. The fractional anisotropy of these phantoms was quantified to compare with measurements made in soft tissues. In addition, soft tissues are also viscoelastic, and we have developed a method to model viscoelastic transversely isotropic materials with the finite element method (FEM). The viscoelastic property estimation from phantom experiments and FEM simulations will also be discussed.

10:30

2aBA8. Applications of acoustic radiation force for quantitative elasticity evaluation of bladder, thyroid, and breast. Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 1st St. SW, Rochester, MN 55905, fatemi@mayo.edu)

Acoustic radiation force (ARF) provides a simple and yet non-invasive mechanism to induce a localized stress inside human body. The response to this excitation is used to estimate the mechanical properties of the targeted tissue *in vivo*. This talk covers an overview of three studies that use ARF for estimation of elastic properties of thyroid, breast, and the bladder in patients. The studies on thyroid and breast were aimed at differentiating between malignant and benign nodules. The study on the bladder was aimed at indirect evaluation of bladder compliance; hence, only a global measurement was needed. The study on breast showed that 16 out of 18 benign masses and 21 out of 25 malignant masses were correctly identified. The study on 9 thyroid patients with 7 benign and 2 malignant nodules showed all malignant nodules were correctly classified and only 2 of the 7 benign nodules were misclassified. The bladder compliance study revealed a high correlation between our method and independent clinical measurement of compliance (R-squared of 0.8–0.9). Further investigations on larger groups of patients are needed to fully evaluate the performances of the methods.

10:50

2aBA9. Multiband center-frequency estimation for robust speckle tracking applications. Emad S. Ebbini and Dalong Liu (Elec. and Comput. Eng., Univ. of Minnesota, 200 Union St. SE, Minneapolis, MN 55455, ebbin001@umn.edu)

Speckle tracking is widely used for the detection and estimation of minute tissue motion and deformation with applications in elastography, shear-wave imaging, thermography, etc. The center frequency of the echo data within the tracking window is an important parameter in the estimation of the tissue displacement. Local variations in this quantity due to echo mixtures (specular and speckle components) may produce a bias in the estimation of tissue displacement using correlation-based speckle tracking methods. We present a new algorithm for estimation and tacking of the center frequency variation in pulse-echo ultrasound as a quantitative tissue property and for robust speckle tracking applications. The algorithm employs multiband analysis in the determination of echo mixtures as a preprocessing step before the estimation of the center frequency map. This estimate, in turn, is used to improve the robustness of the displacement map produced by the correlation-based speckle tracking. The performance of the algorithm is demonstrated in two speckle tracking applications of interest in medical ultrasound: (1) ultrasound thermography and (2) vascular wall imaging.

11:10

2aBA10. Echo decorrelation imaging for quantification of tissue structural changes during ultrasound ablation. T. Douglas Mast, Tyler R. Fosnight, Fong Ming Hooi, Ryan D. Keil, Swetha Subramanian, Anna S. Nagle (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu), Marepalli B. Rao (Environ. Health, Univ. of Cincinnati, Cincinnati, OH), Yang Wang, Xiaoping Ren (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Syed A. Ahmad (Surgery, Univ. of Cincinnati, Cincinnati, OH), and Peter G. Barthe (Guided Therapy Systems/Ardent Sound, Mesa, AZ)

Echo decorrelation imaging is a pulse-echo method that maps millisecond-scale changes in backscattered ultrasound signals, potentially providing real-time feedback during thermal ablation treatments. Decorrelation between echo signals from sequential image frames is spatially mapped and temporally averaged, resulting in images of cumulative, heat-induced tissue changes. Theoretical analysis indicates that the mapped echo decorrelation parameter is equivalent to a spatial decoherence spectrum of the tissue reflectivity, and also provides a method to compensate decorrelation artifacts caused by tissue motion and electronic noise. Results are presented from experiments employing 64-element linear arrays that perform bulk thermal ablation, focal ablation, and pulse-echo imaging using the same piezoelectric elements, ensuring co-registration of ablation and image planes. Decorrelation maps are shown to correlate with ablated tissue histology, including vital staining to map heat-induced cell death, for both ex vivo ablation of bovine liver tissue and *in vivo* ablation of rabbit liver with VX2 carcinoma. Receiver operating characteristic curve analysis shows that echo decorrelation predicts local ablation with greater success than integrated backscatter imaging. Using artifact-compensated echo decorrelation maps, heating-induced decoherence of tissue scattering media is assessed for *ex vivo* and *in vivo* ultrasound ablation by unfocused and focused beams.

11:30

2aBA11. Quantitative ultrasound imaging to monitor *in vivo* high-intensity ultrasound treatment. Goutam Ghoshal (Res. and Development, Acoust. MedSystems Inc., 208 Burwash Ave., Savoy, IL 61874, ghoshal2@gmail.com), Jeremy P. Kemmerer, Chandra Karunakaran, Rami Abuhabshah, Rita J. Miller, and Michael L. Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The success of any minimally invasive treatment procedure can be enhanced significantly if combined with a robust noninvasive quantitative imaging modality. Quantitative ultrasound (QUS) imaging has been widely investigated for monitoring various treatment responses such as chemotherapy and thermal therapy. Previously we have shown the feasibility of using spectral based quantitative ultrasound parameters to monitor high-intensity focused ultrasound (HIFU) treatment of in situ tumors [Ultrasonic Imaging, 2014]. In the present study we examined the use the various QUS parameters to monitor HIFU treatment of an *in vivo* mouse mammary adenocarcinoma model. Spectral parameters in terms of the backscatter coefficient, integrated backscattered energy, attenuation coefficient, and effective scatterer size and concentration were estimated from radiofrequency signals during the treatment. The characteristic of each parameter was compared to the temperature profile recorded by needle thermocouple inserted into the tumor a few millimeters away from the focal zone of the intersecting HIFU and the imaging transducer beams. The changes in the QUS parameters during the HIFU treatment followed similar trends observed in the temperature readings recorded from the thermocouple. These results suggest that QUS techniques have the potential to be used for non-invasive monitoring of HIFU exposure.

11:50

2aBA12. Rapid simulations of diagnostic ultrasound with multiple-zone receive beamforming. Pedro Nariyoshi and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

Routines are under development in FOCUS, the “Fast Object-oriented C++ Ultrasound Simulator” (http://www.egr.msu.edu/~fultras-web), to accelerate B-mode image simulations by combining the fast nearfield method with time-space decomposition. The most recent addition to the FOCUS simulation model implements receive beamforming in multiple zones. To demonstrate the rapid convergence of these simulations in the nearfield region, simulations of a 192 element linear array with an electronically translated 64 element sub-aperture are evaluated for a transient excitation pulse with a center frequency of 3 MHz. The transducers in this simulated array are 5 mm high and 0.5133 mm wide with a 0.1 mm center to center spacing. The simulation is evaluated for a computer phantom with 100,000 scatterers. The same configuration is simulated in Field II (http://field-ii.dk), and the impulse response approach with a temporal sampling rate of 1 GHz is used as reference. Simulations are evaluated for the entire B-mode image simulated with each approach. The results show that, with sampling frequencies of 15 MHz and higher, FOCUS eliminates all of the numerical artifacts that appear in the nearfield region of the B-mode image, whereas Field II requires much higher temporal sampling frequencies to obtain similar results. [Supported in part by NIH Grant R01 EB012079.]

Session 2aED

Education in Acoustics: Undergraduate Research Exposition (Poster Session)

Uwe J. Hansen, Chair
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All posters will be on display from 9:00 a.m. to 11:00 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:00 a.m.

Contributed Papers

2aED1. Prediction of pressure distribution between the vocal folds using Bernoulli's equation. Alexandra Maddox, Liran Oren, Sid Khosla, and Ephraim Gutmark (Univ. of Cincinnati, 3317 Bishop St., Apt. 312, Cincinnati, OH 45219, maddoxat@mail.uc.edu)

Determining the mechanisms of self-sustained oscillation of the vocal folds requires characterization of intraglottal aerodynamics. Since most of the intraglottal aerodynamics forces cannot be measured experimentally, most of the current understanding of vocal fold vibration mechanism is derived from analytical and computational models. Several of such studies have used the Bernoulli's equation in order to calculate the pressure distribution between the vibrating folds. In the current study, intraglottal pressure measurements are taken in a hemilarynx model and are compared with pressure values that are computed form the Bernoulli's equation. The hemilarynx model was made by removing one fold and having the remaining fold vibrating against a metal plate. The plate was equipped with two pressure ports located near the superior and inferior aspects of the fold. The results show that pressure calculated using Bernoulli's equation matched well with the measured pressure waveform during the glottal opening phase and dissociated during the closing phase.

2aED2. Effects of room acoustics on subjective workload assessment while performing dual tasks. Brenna N. Boyd, Zhao Peng, and Lily Wang (Eng., Univ. of Nebraska at Lincoln, 11708 s 28th St., Bellevue, NE 68123, bnboyd@unomaha.edu)

This investigation examines the subjective workload assessments of individuals using the NASA Task Load Index (TLX), as they performed speech comprehension tests under assorted room acoustic conditions. This study was motivated due to the increasing diversity in US classrooms. Both native and non-native English listeners participated, using speech comprehension test materials produced by native English speakers in the first phase and by native Mandarin Chinese speakers in the second phase. The speech materials were disseminated in an immersive listening environment to each listener under 15 acoustic conditions, from combinations of background noise level (three levels from RC-30, 40, and 50) and reverberation time (five levels from 0.4 to 1.2 seconds). During each condition, participants completed assorted speech comprehension tasks while also tracing a moving dot for an adaptive rotor pursuit task. At the end of each acoustic condition, listeners were asked to assess the perceived workload by completing the six-item NASA TLX survey, e.g., mental demand, perceived performance, effort, and frustration. Results indicate that (1) listeners' workload assessments degraded as the acoustic conditions became more adverse, and (2) the decrement in subjective assessment was greater for non-native listeners.

2aED3. Analysis and virtual modification of the acoustics in the Nebraska Wesleyan University campus theatre auditorium. Laura C. Brill (Dept. of Phys., Nebraska Wesleyan Univ., 5000 St. Paul Ave, Lincoln, NE 68504, lbrill@nebrwesleyan.edu), Matthew G. Blevins, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE)

NWU's McDonald Theatre Auditorium is used for both musical and non-musical performances. The acoustics of the space were analyzed in order to determine whether the space could be modified to better fit its uses. The acoustic characteristics of the room were obtained from impulse responses using the methods established in ISO 3382-1 for measuring the acoustic parameters of a performance space. A total of 22 source/receiver pairs were used. The results indicate a need for increased reverberation in the mid to high frequency ranges of 500–8000 Hz. The experimental results were used to calibrate a virtual model of the space in ODEON acoustics software. Materials in the model were then successfully modified to increase reverberation time and eliminate unwanted flutter echoes to optimize the acoustics to better suit the intended purposes of the space.

2aED4. The diffraction pattern associated with the transverse cusp caustic. Carl Frederickson and Nicholas L. Frederickson (Phys. and Astronomy, Univ. of Central Arkansas, LSC 171, 201 Donaghey Ave., Conway, AR 72035, nicholaslfrederickson@gmail.com)

New software has been developed to evaluate the Pearcey function $P_{\pm}(w_1,w_2)=\int_{(-\infty)}^{\infty} \exp[\pm i(s^{4/4} + w_2 s^{2/2} + w_1 s)]ds$. This describes the diffraction pattern of a transverse cusp caustic. Run-time comparisons between different coding environments will be presented. The caustic surface produced by the reflection of a spherical wavefront from the surface given by $h(x,y) = h_{1x}^2 + h_{2xy}^2 + h_{3y}^2$ will also be displayed.

2aED5. Architectural acoustical oddities. Zev C. Woodstock and Caroline P. Lubert (Mathematics & Statistics, James Madison Univ., 301 Dixie Ave., Harrisonburg, VA 22801, lubertcp@jmu.edu)

The quad at James Madison University (Virginia, USA) exhibits an uncommon, but not unique, acoustical oddity called Repetition Pitch. When someone stands at certain places on the quad and makes a punctuated white noise (claps, for example) a most unusual squeak is returned. This phenomenon only occurs at these specific places. A similar effect has been observed in other locations, mostly notably Ursinus College (Pennsylvania, USA) and the pyramid at Chichen Itza (Mexico). This talk will discuss Repetition Pitch, as well as other interesting architectural acoustic phenomenon including the noisy animals in the caves at Arcy-sur-Cure (France), the early warning system at Golkonda Fort (Southern India) and the singing angels at Wells Cathedral in the United Kingdom.

2aED6. Impedance tube measurements of printed porous materials. Carl Frederickson and Forrest McDougal (Phys. and Astronomy, Univ. of Central Arkansas, LSC 171, 201 Donaghey Ave., Conway, AR 72035, FMCDOUGAL1@CUB.UCA.EDU)

An impedance tube has been used to make measurements of the acoustic impedance of porous samples. Porous with designed porosities and tortuosities have been produced using 3D printing. Measured impedances are compared to calculated values.

2aED7. Stick bombs: A study of the speed at which a woven stick construction self-destructs. Scotty McKay and William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, SMCKAY2@uca.edu)

A stick bomb is created by weaving sticks together in a particular pattern. By changing the way the sticks are woven together, different types of stick bombs are created. After the stick bomb is woven to the desired length, one side of the stick bomb can be released causing it to rapidly begin tearing itself apart in the form of a pulse that propagates down the weave. This occurs due to a large amount of potential energy stored within the multitude of bent sticks; however, the physics of this phenomena has not been studied to the authors knowledge. The linear mass density of the stick bomb can be changed by varying the tightness of the weave. Data on these stick bombs, including video analysis to determine the pulse speed, will be presented.

2aED8. Three-dimensional printed acoustic mufflers and aeroacoustic resonators. John Ferrier and William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, jpferrierjr@gmail.com)

We explore and present the use of 3D printing technology to design, construct, and test acoustic elements that could be used as a low-frequency Helmholtz-resonator style muffler in a ventilation duct system. Acoustic elements such as these could be quickly prototyped, printed, and tested for any noisy duct environment. These acoustic elements are tested with and without mean flow to characterize their sound absorption (and sound generation) properties. It is found that at particular ranges of air flow speeds the simply designed acoustic muffler acts as a site for aeroacoustic sound generation. Measurement data and 3D model files with Python-scripting will be presented for several muffler designs. This work is supported by the Arkansas Space Grant Consortium in collaboration with NASA's Acoustics Office at the Johnson Space Center.

2aED9. Determining elastic moduli of concrete using resonance. Gerard Munyazikwiye and William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, GMUNYAZIKWIYE1@uca.edu)

The elastic moduli of rods of material can be determined by resonance techniques. The torsional, longitudinal, and transverse resonance modes for a rod of known mass and length can be measured experimentally. These resonance frequencies are related to the elastic properties of the material, hence, by measuring these quantities the strength of the material can be determined. Preliminary tests as proof of principle are conducted with metallic rods. Data and experimental techniques for determining the elastic moduli for concrete using this procedure will be presented.

2aED10. Articulation of sibilant fricatives in Colombian Spanish. Alexandra Abell and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 404 West Kirkwood Ave., Bloomington, IN 47404, alabell@indiana.edu)

Colombians constitute the largest South American population in the United States at 909,000 (or 24% of the total South American population in the U.S.), and Bogotá, Colombia is the most populated area within the Andean Highland region, yet relatively little is known about Colombian Spanish speech production. The majority of previous studies of Colombian phonetics have relied on perception and acoustic analysis. The present study contributes to Colombian Spanish phonetics by investigating the articulation of sibilant fricatives. In particular, the shape of the palate and tongue during the production of sibilants is investigated in an attempt to quantify the shape and size of the oral cavity in the vicinity of the sibilant constriction. Real-time three-dimensional ultrasound, palate impressions, acoustic recordings, and electroglottography are brought to bear on these issues.

2aED11. Teaching acoustical interaction: An exploration of how teaching architectural acoustics to students spawns project-based learning. Daniel Butko, Haven Hardage, and Michelle Oliphant (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, Haven.B.Hardage-1@ou.edu)

The language and methods of architecture typically evaluated through small-scale models and drawings can be complemented by full-scale interactive constructs, augmenting learning through participatory, experiential, and sometimes experimental means. Congruent with Constantin Brancusi's proclamation, "architecture is inhabitable sculpture," opportunities to build full-scale constructs introduce students to a series of challenges predicated by structure, connections, safety, and a spirit of inquisition to learn from human interaction. To educate and entertain through sensory design, undergraduate students designed and built an interactive intervention allowing visual translation of acoustical impulses. The installation was developed and calibrated upon the lively acoustics and outward campus display of the college's gallery, employing excessive reverberation and resonance as a method of visually demonstrating sound waves. People physically inhabiting the space were the participants and critics by real-time reaction to personal interaction. The learning process complemented studio-based instruction through hands-on interaction with physical materials and elevated architectural education to a series of interactions with people. This paper documents and celebrates the Interactive Synchronicity project as a teaching tool outside common studio project representation while enticing classmates, faculty, and complete strangers to interact with inhabitable space.

2aED12. Palate shape and the central tongue groove. Coretta M. Talbert (Speech and Hearing Sci., Univ. of Southern MS, 211 Glen Court, Jackson, MS 39212, coretta.talbert@eagles.usm.edu) and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

It is well known that the center of the tongue can be grooved so that it is lower in the mouth than the lateral parts of the tongue, or it can bulge higher than the lateral parts of the tongue. It has never been shown whether or how this groove or bulge is related to the shape of the palate. In this study, we investigated the shape and size of the palate for several speakers using digitized 3D laser-scans of palate impressions and measurements on the impression plasters themselves. The groove or bulge in the center of the tongue was measured using real-time three-dimensional ultrasound. Pertinent findings will be presented concerning the relationship of the central groove/bulge shape and size to the shape and size of the palate.

2aED13. Signal processing for velocity and range measurement using a micromachined ultrasound transducer. Dominic Guri and Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Anderson 204, Medford, MA 02155, dominic.guri@tufts.edu)

Signal processing techniques are under investigation for determination of range and velocity information from MEMS based ultrasound transducers. The ideal technique will be real-time, result in high resolution and accurate measurements, and operate successfully in noise. Doppler velocity measurements were previously demonstrated using a MEMS cMUT array (Shin *et al*., ASA Fall Meeting 2011, JASA 2013, Sens. Actuators A 2014). The MEMS array has 168 nickel-on-glass capacitive ultrasound transducers on a 1 cm die, and operates at 180 kHz in air. Post processing of the received ultrasound demonstrated the ability to sense velocity using continuous wave (CW) Doppler at a range of up to 1.5 m. The first attempt at real-time processing using a frequency modulated continuous wave (FM/CW) scheme was noise limited by the analog demodulation circuit. Further noise analysis is ongoing to determine whether this scheme may be viable. Other schemes under consideration include cross correlation chirp and single and multi-frequency burst waveforms. Preliminary results from a single frequency burst showed that cross-correlation-based signal processing may achieve acceptable range. The system is targeted at short range small robot navigation tasks. Determination of surface roughness from scattering of the reflected waves may also be possible.

2aED14. Investigation of a tongue-internal coordinate system for two-dimensional ultrasound. Rebecca Pedro, Elizabeth Mazzocco (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, rebpedro@indiana.edu), Tamás G. Csapó (Dept. of Telecommunications and Media Informatics, Budapest Univ. of Technol. and Economics, Budapest, Hungary), and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

In order to compare ultrasound recordings of tongue motion across utterances or across speakers, it is necessary to register the ultrasound images with respect to a common frame of reference. Methods for doing this typically rely either (1) on fixing the position of the ultrasound transducer relative to the skull by means of a helmet or a similar device, or (2) re-aligning the images by various means, such as optical tracking of head and transducer motion. These methods require sophisticated laboratory setups, and are less conducive to fieldwork or other studies in which such methods are impractical. In this study, we investigated the possibility of defining a rough coordinate system for image registration based on anatomical properties of the tongue itself. This coordinate system is anchored to the lower-jaw rather than the skull, but may potentially be transformed into an approximately skull-relative coordinate system by integrating video recordings of jaw motion.

2aED15. The effect of finite impedance ground reflections on horizontal full-scale rocket motor firings. Samuel Hord, Tracianne B. Neilsen, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., 737 N 600 E #103, Provo, UT 84606, samuel.hord@gmail.com)

Ground reflections have a significant impact on the propagation of sound from a horizontal rocket firing. The impedance of the ground relies strongly on effective flow resistivity of the surface and determines the frequencies at which interference nulls occur. For a given location, a softer ground, with lower effective flow resistivity, shifts the location of interference nulls to lower frequencies than expected for a harder ground. The difference in the spectral shapes from two horizontal firings of GEM-60 rocket motors, over snowy ground, clearly shows this effect and has been modeled. Because of the extended nature of high energy launch vehicles, the exhaust plume is modeled as a partially correlated line source, with distribution parameters chosen to match the recorded data sets as best as possible. Different flow resistivity values yield reasonable comparisons to the results of horizontal GEM-60 test firings.

2aED16. Palate-related constraints on sibilant production in three dimensions. Sarah Janssen and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, sejansse14@gmail.com)

Most studies of speech articulation are limited to a single plane, typically the midsagittal plane, although coronal planes are also used. Single-plane data have been undeniably useful in improving our understanding of speech production, but for many acoustic and aerodynamic processes, a knowledge of 3D vocal tract shapes is essential. In this study, we used palate impressions to investigate variations in the 3D structure of the palates of several individuals, and we used real-time 3D ultrasound to image the tongue surface during sibilant production by the same individuals. Our analysis focused on the degree to which tongue shapes during sibilant productions are substantially similar or different between individuals with different palate shapes and sizes.

2aED17. The evaluation of impulse response testing in low signal-to-noise ratio environments. Hannah D. Knorr (Audio Arts and Acoust., Columbia College Chicago, 134 San Carlos Rd, Address, Minooka, IL 60447, hknorr13@gmail.com), Jay Bleifnick (Audio Arts and Acoust., Columbia College Chicago, Schiller Park, IL), Andrew M. Hulva, and Dominique J. Chéenne (Audio Arts and Acoust., Columbia College Chicago, Chicago, IL)

Impulse testing is used by industry professionals to test many parameters of room acoustics, including the energy decay, frequency response, time response, etc. Current testing software makes this process as streamlined as possible, but generally must be utilized in quiet environments to yield high signal-to-ratios and more precise results. However, many real world situations cannot conform to the necessary standards needed for reliable data. This study tests various methods of impulse responses in background noise environments in an attempt to find the most reliable procedure for spaces with a high ambient noise levels. Additionally, extreme situations will be evaluated and a method will be derived to correct for the systematic error attributed to high background noise levels.

2aED18. Comparison of palate impressions and palate casts from three-dimensional laser-scanned digital models. Michelle Tebout and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, mtebout@imail.iu.edu)

Palate impressions and their casts in plaster are negatives of each other. While plaster casts are the standard for palate measurements and data preservation, making such casts can be time-consuming and messy. We hypothesized that measurements from 3D laser-scanned palate impressions are negligibly different from equivalent measurements from 3D laser-scanned palate casts. If true, this would allow the step of setting impressions in plaster to be skipped in future research. This poster presents the results of our study.

2aED19. The analysis of sound wave scattering using a firefighter's Personal Alert Safety System signal propagating through a localized region of fire. Andrew L. Broda (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402), Chase J. Rudisill (Phys. Dept., U.S. Naval Acad., Harwood, MD), Nathan D. Smith (Phys. Dept., U.S. Naval Acad., Davidsonville, MD), Matthew K. Schrader, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD, korman@usna.edu)

Firefighting is quite clearly a dangerous and risk-filled job. To combat these dangers and risks, firefighters wear a (National Fire Protection Agency, NFPA 2007 edition of the 1982 standard) Personal Alert Safety System (PASS) that will sound a loud alarm if it detects (for example) the lack of movement of a firefighter. However, firefighters have experienced difficulty locating the source of these alarm chirps (95 dBA around 3 kHz) in a burning building. The project goal is to determine the effect of pockets of varying temperatures of air in a burning building on the sound waves produced by a PASS device. Sound scattering experiments performed with a vertical heated air circular jet plume (anechoic chamber) and with a wood fire plume from burning cylindrical containers (Anne Arundel Fire Department's Training Facility) suggest that from Snell's Law, sound rays refract around such pockets of warmer air surrounded by less warmer ambient air due to changes in the sound speed with temperature through the medium. Real-time and spectral measurements of 2.7 kHz CW sound scattering (using a microphone) exhibit some attenuation and considerable amplitude and frequency modulation. This research may suggest future experiments and effective modifications of the current PASS system.

2aED20. New phased array models for fast nearfield pressure simulations. Kenneth Stewart and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, stewa584@msu.edu)

FOCUS, the "Fast Object-oriented C++ Ultrasound Simulator," is free MATLAB-based software that rapidly and accurately models therapeutic and diagnostic ultrasound with the fast nearfield method, time-space decomposition, and the angular-spectrum approach. FOCUS presently supports arrays of circular, rectangular, and spherically focused transducers arranged in flat planar, spherically focused, and cylindrically focused geometries. Excellent results are obtained with all of these array geometries in FOCUS for simulations of continuous-wave and transient excitations, and new array geometries are needed for B-mode simulations that are presently under development. These new array geometries also require new data structures that describe the electrical connectivity of the arrays. Efforts to develop these new features in FOCUS are underway, and results obtained with these new array geometries will be presented. Other new features for FOCUS will also be demonstrated. [Supported in part by NIH Grant R01 EB012079.]

2aED21. Nonlinear scattering of crossed focused ultrasonic beams in the presence of turbulence generated behind a model deep vein thrombosis using an orifice plate set in a thin tube. Daniel Fisher and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

An orifice plate (modeling a "blockage" in a deep vein thrombosis DVT) creates turbulent flow in a downstream region of a submerged polyethylene tube (1.6 mm thick, diameter 4 cm and overall length 40 cm). In the absence of the orifice plate, the water flow is laminar. The orifice plate is mechanically secured between two 20 cm tube sections connected by a union. The union allows a plate with an orifice to be slid into the union providing a concentric orifice plate that can obstruct the flow causing vorticity and turbulent flow downstream. A set of orifice plates (3 mm thick) are used (one at a time) to conveniently obstruct the tube flow with a different radius compared to the inner wall tube radius. The nonlinear scattering at the sum frequency (f_+ = 3.8 MHz), from mutually perpendicular spherical focused beams (f_1 = 1.8 MHz and f_2 = 2.0 MHz) is used to correlate the Doppler shift, spectral, and intensity as a function of the orifice plate size in an effort to correlate the blockage with the amount of nonlinear scattering. In the absence of turbulence in the overlap region, there is virtually no scattering. Therefore, a slight blockage is detectable.

2aED22. Analysis of acoustic data acquisition instrumentation for underwater blast dredging. Brenton Wallin, Alex Stott, James Hill, Timothy Nohara, Ryan Fullan, Jon Morasutti, Brad Clark, Alexander Binder, and Michael Gardner (Ocean Eng., Univ. of Rhode Island, 30 Summit Ave., Narragansett, RI 02882, brentwallin@my.uri.edu)

A team of seniors from the University of Rhode Island were tasked with analyzing the acoustic data and evaluating the data acquisition systems used in Pacific Northwest National Laboratories' (PNNL) study of blast dredging in the Columbia River. Throughout the semester, the students learned about the unique acoustic signatures of confined underwater blasts and the necessary specifications of systems used to record them. PNNL used two data acquisition systems. One was a tourmaline underwater blast sensor system created by PCB Piezotronics. The second was a hydrophone system using a Teledyne TC 4040 hydrophone, a Dytran inline charge amplifier, and a signal conditioner built for the blast sensor system. The students concluded that the data from the blast sensor system was reliable because the system was built by the company for this specific application and there were calibration sheets showing the system worked properly. The hydrophone data was deemed unreliable because components were orientated in an unusual manner that lead to improper data acquisition. A class of URI graduate students built a new hydrophone system that accurately recorded underwater dredge blasts performed in New York Harbor. This system is a fraction of the price of the blast sensor system.

2aED23. Effects of sustainable and traditional building systems on indoor environmental quality and occupant perceptions. Joshua J. Roberts and Lauren M. Ronsse (Audio Arts and Acoust., Columbia College Chicago, 4363 N. Kenmore Ave., Apt. #205, Chicago, IL 60613, joshua.roberts@loop.colum.edu)

This study examines the effects of both sustainable and traditional building systems on the indoor environmental quality (IEQ) and occupant perceptions in an open-plan office floor of a high-rise building located in Chicago, IL. The office evaluated has sustainable daylighting features as well as a more traditional variable air volume mechanical system. Different measurement locations and techniques are investigated to quantify the indoor environmental conditions (i.e., acoustics, lighting, and thermal conditions) experienced by the building occupants. The occupant perceptions of the indoor environmental conditions are assessed via survey questionnaires administered to the building occupants. The relationships between the IEQ measured in the office and the occupant perceptions are assessed.

TUESDAY MORNING, 28 OCTOBER 2014 — MARRIOTT 9/10, 8:00 A.M. TO 12:15 P.M.

Session 2aID

Archives and History and Engineering Acoustics: Historical Transducers

Steven L. Garrett, Chair
Grad. Prog. in Acoustics, Penn State, Applied Research Lab, P. O. Box 30, State College, PA 16804-0030

Chair's Introduction—8:00

Invited Papers

8:05

2aID1. 75th Anniversary of the Shure Unidyne microphone. Michael S. Pettersen (Applications Eng., Shure Inc., 5800 W. Touhy Ave., Niles, IL 60714, pettersen_michael@shure.com)

2014 marks the 75th anniversary of the Shure model 55 microphone. Introduced in 1939 and still being manufactured today, the Shure Unidyne was the first unidirectional microphone using a single dynamic element. The presentation provides an overview of the Unidyne's unique position in the history of 20th century broadcast, politics, and entertainment, plus the amazing story of Benjamin Bauer, a 24 year old immigrant from the Ukraine who invented the Unidyne and earned his first of over 100 patents for audio technology. Rare Unidyne artifacts from the Shure Archive will be on display after the presentation, including prototypes fabricated by Ben Bauer.

8:25

2aID2. Ribbon microphones. Wesley L. Dooley (Eng., Audio Eng. Assoc., 1029 North Allen Ave, Pasadena, CA 91104, wes@ribbonmics.com)

The ribbon microphone was invented by Dr. Walter Schottky who described it in German Patent 434855C, issued December 21, 1924 to Siemens & Halske (S&H) in Berlin. An earlier "Electro-Dynamic Loudspeaker" Patent which Schottky had written with Dr. Erwin Gerlach described a compliant, lightweight, and ribbed aluminum membrane whose thinnest dimension was at right angles to a strong magnetic field. Passing an audio frequency current through this membrane causes it to move and create sound vibrations. The December Patent describes how this design functions either as a loudspeaker or a microphone. A 1930 S&H patent for ribbon microphone improvements describes how they use internal resonant and ported chambers to extend frequency response past 4 kHz. RCA dramatically advanced ribbon microphone performance in 1931. They opened the ribbon to free air to create a consistent, air-damped, low-distortion, figure-eight with smooth 30–10,000 Hz response. RCA ribbon microphones became the performance leader for cinema, broadcast, live sound and recording. Their 20–30,000 Hz RCA 44B and BX was manufactured from 1936 to 1955. It is the oldest design still used every day at major studios. Ribbon microphones are increasingly used for contemporary recordings. Come hear why ribbon microphones, like phonograph records, are relevant to quality sound.

8:45

2aID3. Iconic microphonic moments in historic vocal recordings. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Microphone selection—the strategic pairing of microphone make and model with each sound to be recorded—is one of the most important decisions a sound engineer must make. The technical specifications of the microphone identify which transducers are capable of functioning properly for any given recording task, but the ultimate decision is a creative one. The goal is for the performance capabilities of the microphone to not only address any practical recording session challenges, but also flatter the sound of the instrument, whether in pursuit of palpable realism or a fictionalized new timbre. The creative decision is informed, in part, by demonstrated success in prior recordings, the most important of which are described for that essential pop music instrument: the voice.

9:05

2aID4. The WE 640AA condenser microphone. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com)

In 1916 Edward Wente, working for Western Electric an AT&T subsidiary, invented a microphone that was the foundation of the modern condenser microphone. Wente's early condenser microphone designs continued to be developed until Western Electric produced the WE 361 in 1924 followed by the Model 394 condenser microphone in 1926. Western Electric used the WE 394 microphone as part of the "Master Reference System" to rate audio transmission quality of the telephone network. The WE 394 was too large for some measurement purposes so in 1932 Bell Labs engineers H. Harrison and P. Flanders designed a smaller version. The diaphragm had a diameter of 0.6 in. However, this design proved too difficult to manufacture and F. Romanow, also at Bell Labs, designed the 640A "1 in." microphone in 1932. Years later it was discovered that the 640A sensitivity varied by almost 6 dB from -650 C to 250 C. To reduce the thermal sensitivity, Bell Labs engineers M. Hawley and P. Olmstead carefully changed some of the 640A materials. The modified microphone was designated as the 640AA, which became the worldwide standard microphone for measuring sound pressure. This talk will describe some more details of the history of the 640AA microphone.

9:25

2aID5. Reciprocity calibration of condenser microphones. Leo L. Beranek (Retired, 10 Longwood Dr., Westwood, MA 02090, beranekleo@ieee.org)

The theory of reciprocity began with Lord Rayleigh and was first well stated by S. Ballantine (1929). The first detailed use of the reciprocity theory for the calibration of microphones was by R. K. Cook (1940). At the wartime Electro-Acoustic Laboratory, at Harvard University, the need arose to calibrate a large number of Western Electric 640-AA condenser microphones. A reciprocity apparatus was developed that connected the two microphones with an optimum shaped cavity that included a means for introducing hydrogen or helium to extend the frequency range. The apparatus was published by A. L. Dimattia and F. M. Wiener (1946). A number of things resulted. The Harvard group, in 1941, found that the international standard of sound pressure was off by 1.2 dB—that standard was maintained by the French Telephone Company and the Bell Telephone Laboratories and was based on measurements made with Thermophones. This difference was brought to the attention of those organizations and the reciprocity method of calibration was subsequently adopted by them resulting in the proper standard of sound pressure adopted around 1942. The one-inch condenser microphone has subsequently become the worldwide standard for precision measurement of sound field pressures.

9:45–10:00 Break

10:00

2aID6. Electret microphones. James E. West (ECE, Johns Hopkins Univ., 3400 N. Charles St., Barton Hall 105, Baltimore, MD 21218, jimwest@jhu.edu)

For nearly 40 years, condenser electret microphones have been the transducer of choice in most every area of acoustics including telephony, professional applications, hearing aids, and toys. More than 2 billion electret microphones are produced annually, primarily for the communications and entertainment markets. E. C. Wente invented the condenser microphone in 1917 at Bell Labs while searching for a replacement for the carbon microphone used in telephones; however, the necessary few hundred volt bias rendered the condenser microphone unusable in telephony, but its acoustical characteristics were welcomed in professional and measurement applications. Permanently charged polymers (electrets) provided the necessary few hundred-volt bias, thus simplifying the mechanical and electrical requirements for the condenser microphone and making it suitable for integration into the modern telephone. The introduction of inexpensive condenser microphones with matching frequency, phase, and impedance characteristics opened research opportunities for multiple microphone arrays. Array technology developed at Bell Labs will be presented in this talk.

10:20

2aID7. Meucci's telephone transducers. Angelo J. Campanella (Acculab, Campanella Assoc., 3201 Ridgewood Dr., Ohio, Hilliard, OH 43026, a.campanella@att.net)

Antonio Meucci (1809–1889) developed variable reluctance transducers from 1854 to 1876 and beyond after noticing that he could hear voice sounds from paddle electrodes while he participated in electrotherapy of a migraine patient around 1844 while in Havana, Cuba. He immigrated to Staten Island, NY, in 1850 and continued experimenting to develop a telephone. He found better success from electromagnetics using materials evolved from the telegraph developed by Morse, as well as a non-metal diaphragm with an iron tab, iron bars and a horseshoe shape. Artifacts from his residence on Staten Island are presently on display at the museum of his life on Staten Island from 1850 to his death. Those artifacts, thought until now to be only models, were found to be wired and still operative. Tests were performed in July, 2011. Their electrical resistance is that expected for wire wound variable reluctance transducers. Voice signals were produced without any externally supplied operating current. At least one transducer was found to be also operable as a receiver and was driven to produce voice sounds to the ear. Meucci's life and works will be discussed and these test results will be demonstrated including recordings from voice tests.

10:40

2aID8. The Fessenden Oscillator: The first sonar transducer. Thomas R. Howarth and Geoffrey R. Moss (U.S. Navy, 1176 Howell St, B1346 R404A, Newport, RI 02841, thomas.howarth@navy.mil)

When the RMS Titanic sunk in 1912, there was a call placed forth by ship owners for inventors to offer solutions for ship collision avoidance methods. Canadian born inventor Reginald A. Fessenden answered this call while working at the former Boston Submarine Signal Company with the invention and development of the first modern transducer used in a sonar. The Fessenden oscillator was an edge clamped circular metal with a radiating head facing the water on one side while the interior side had a copper tube attached that moved in and out of a fixed magnetic coil. The coil consisted of a direct-current (DC) winding to provide a magnetic field polarization and an alternating-current (AC) coil winding to induce the current into the copper tube and thus translate the magnetic field polarization to the radiating plate with vibrations that translated from the radiating head to the water medium. The prototype and early model versions operated at 540 Hz. Later developments included adaptation of this same transducer for use in underwater communications, obstacle avoidance with WW I retrofits onto British submarines for both transmitting and receiving applications including mine detection. This presentation will discuss design details including a modern numerical modelling effort.

11:00

2aID9. Historical review of underwater acoustic cylindrical transducer development in Russia for sonar arrays. Boris Aronov (ATMC/ECE, Univ. of Massachusetts Dartmouth, Needam, MA) and David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

Beginning with the introduction of piezoelectric ceramics in the 1950's, underwater acoustics transducer development for active sonar arrays proceeded in different directions in Russia (formerly USSR) than in the United States (US). The main sonar arrays in Russia were equipped with cylindrical transducers, whereas in the US, the implementation was most often made with extensional bar transducers of the classic Tonpilz design. The presentation focuses on the underlying objectives and humane factors that shaped the preference towards the widespread application of baffled cylindrical transducers for arrays in Russia, the history of their development, and contributions to theory of the transducers made by the pioneering developers.

11:20

2aID10. The phonodeik: Measuring sound pressure before electroacoustic transducers. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16802, sct12@psu.edu)

The modern ability to visualize sound pressure waveforms using electroacoustic transducers began with the development of the vacuum tube amplifier, and has steadily improved as better electrical amplification devices have become available. Before electoral amplification was available; however, a significant body of acoustic pressure measurements had been made using the phonodeik, a device developed by Dayton C. Miller in the first decade of the twentieth century. The phonodeik employs acoustomechanical transduction to rotate a small mirror that reflects an optical beam to visualize the pressure waveform. This presentation will review the device and some of the discoveries made with it.

11:40

2aID11. A transducer not to be ignored: The siren. Julian D. Maynard (Phys., Penn State Univ., 104 Davey Lab, Box 231, University Park, PA 16802, maynard@phys.psu.edu)

An historic transducer to which one should pay attention is the siren. While its early application was as a source for a musical instrument, the siren soon became the transducer of choice for long-range audible warning because of its high intensity and recognizable tone. The components defining the siren include a solid stator and rotor, each with periodic apertures, and a compressed fluid (usually air but could be other fluids). With the rotor rotating in close proximity to the stator, and the resulting opening and closing of passageways through the apertures for the compressed fluid results in periodic sound waves in the surrounding fluid; usually a horn is used to enhance the radiation efficiency. The high potential energy of the compressed fluid permits high intensity sound. Some sirens which received scientific study include that of R. Clark Jones (1946), a 50 horsepower siren with an efficiency of about 70%, and that of C. H. Allen and I. Rudnick (1947), capable of ultrasonic frequencies and described as a "supersonic death ray" in the news media. Some design considerations, performance results, and applications for these sirens will be presented.

12:00–12:15 Panel Discussion

Session 2aMU

Musical Acoustics: Piano Acoustics

Nicholas Giordano, Chair
Physics, College of Sciences and Mathematics, Auburn University, Auburn, AL 36849

Invited Papers

9:00

2aMU1. The slippery path from piano key to string. Stephen Birkett (Systems Design Eng., Univ. of Waterloo, 250 University Ave., Waterloo, ON N2L 3G1, Canada, sbirkett@uwaterloo.ca)

Everything that contributes to the excitation of a piano string, from key input to hammer–string interaction, is both deterministic and consistently repeatable. Sequences of identical experimental trials give results that are indistinguishable. The simplicity of this behavior contrasts with the elusive goal of predicting input–output response and the extreme difficulty of accurate physical characterization. The nature and complexity of the mechanisms and material properties involved, as well as the sensitivity of their parameterization, place serious obstacles in the way of the usual investigative tools. This paper discusses and illustrates the limitations of modeling and simulation as applied to this problem, and the special considerations required for meaningful experimentation.

9:25

2aMU2. Coupling between transverse and longitudinal waves in piano strings. Nikki Etchenique, Samantha Collin, and Thomas R. Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, netchenique@rollins.edu)

It is known that longitudinal waves in piano strings noticeably contribute to the characteristic sound of the instrument. These waves can be induced by directly exciting the motion with a longitudinal component of the piano hammer, or by the stretching of the string associated with the transverse displacement. Longitudinal waves that are induced by the transverse motion of the string can occur at frequencies other than the longitudinal resonance frequencies, and the amplitude of the waves produced in this way are believed to vary quadratically with the amplitude of the transverse motion. We present the results of an experimental investigation that demonstrates the quadratic relationship between the magnitude of the longitudinal waves and the magnitude of the transverse displacement for steady-state, low-amplitude excitation. However, this relationship is only approximately correct under normal playing conditions.

9:50

2aMU3. Microphone array measurements, high-speed camera recordings, and geometrical finite-differences physical modeling of the grand piano. Rolf Bader, Florian Pfeifle, and Niko Plath (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

Microphone array measurements of a grand piano soundboard show similarities and differences between eigenmodes and forced oscillation patterns when playing notes on the instrument. During transients the driving point of the string shows enhanced energy radiation, still not as prominent as with the harpsichord. Lower frequencies are radiated stronger on the larger side of the soundboard wing shape, while higher frequencies are radiated stronger on the smaller side. A separate region at the larger part of the wing shape, caused by geometrical boundary conditions has a distinctly separate radiation behavior. High-speed camera recordings of the strings show energy transfer between strings of the same note. In physical models including hammer, strings, bridge, and soundboard the hammer movement is crucially necessary to produce a typical piano sound. Different bridge designs and bridge models are compared enhancing inharmonic sound components due to longitudinal-transversal coupling of the strings at the bridge.

10:15–10:35 Break

10:35

2aMU4. Adjusting the soundboard's modal parameters without mechanical change: A modal active control approach. Adrien Mamou-Mani (IRCAM, 1 Pl. Stravinsky, Paris 75004, France, adrien.mamou-mani@ircam.fr)

How do modes of soundboards affect the playability and the sound of string instruments? This talk will investigate this question experimentally, using modal active control. After identifying modal parameters of a structure, modal active control allows the adjustments of modal frequency and damping thanks to a feedback loop, without any mechanical changes. The potential of this approach for musical acoustics research will be presented for three different instruments: a simplified piano, a guitar, and a cello. The effects of modal active control of soundboards will be illustrated on attack, amplitude of sound partials, sound duration, playability, and "wolf tone" production.

11:00

2aMU5. Modeling the influence of the piano hammer shank flexibility on the sound. Juliette Chabassier (Magique 3D, Inria, 200 Ave. de la vieille tour, Talence 33400, France, juliette.chabassier@inria.fr)

A nonlinear model for a vibrating Timoshenko beam in non-forced unknown rotation is derived from the virtual work principle applied to a system of beam with mass at the end. The system represents a flexible piano hammer shank coupled to a hammer head. A novel energy-based numerical scheme is then provided and coupled to a global energy-preserving numerical solution for the whole piano (strings, soundboard, and sound propagation in the air). The obtained numerical simulations show that the pianistic touch clearly influences the spectrum of the piano sound of equally loud isolated notes. These differences do not come from a possible shock excitation on the structure, nor from a changing impact point, nor a "longitudinal rubbing motion" on the string, since neither of these features are modeled in our study.

Contributed Paper

11:25

2aMU6. Real-time tonal self-adaptive tuning for electronic instruments. Yijie Wang and Timothy Y. Hsu (School of Music, Georgia Inst. of Technol., 950 Marietta St. NW Apt 7303, Atlanta, GA 30318, yijiewang@gatech.edu)

A fixed tuning system cannot achieve just intonation on all intervals. A better approximation of just intonation is possible if the frequencies of notes are allowed to vary. Adaptive tuning is a class of methods that adjusts the frequencies of notes dynamically in order to maximize musical consonance. However, finding the optimal frequencies of notes directly based on some definition of consonance has shown to be difficult and computationally expensive. Instead, this paper proposes that the current key of the music is both a good summary of past notes and a good prediction of future notes, which can facilitate adaptive tuning. A method is proposed that uses a hidden Markov model to detect the current key of the music and compute optimal frequencies of notes based on the current key. In addition, a specialized online machine learning method that enforces symmetry among diatonic keys is presented, which can potentially adapt the model for different genres of music. The algorithm can operate in real time, is responsive to the notes played, and is applicable to various electronic instruments, such as MIDI pianos. This paper also presents comparisons between this proposed tuning system and conventional tuning systems.

TUESDAY MORNING, 28 OCTOBER 2014 MARRIOTT 3/4, 9:25 A.M. TO 11:35 A.M.

Session 2aNSa

Noise and Psychological and Physiological Acoustics: New Frontiers in Hearing Protection I

William J. Murphy, Cochair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998

Elliott H. Berger, Cochair
Occupational Health & Environmental Safety Division, 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

Chair's Introduction—9:25

Invited Papers

9:30

2aNSa1. How long are inexperienced-subjects "naive" for ANSI S12.6? Hilary Gallagher, Richard L. McKinley (Battlespace Acoust. Branch, Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, richard.mckinley.1@us.af.mil), and Melissa A. Theis (ORISE, Air Force Res. Lab., Wright-Patterson AFB, OH)

ANSI S12.6-2008 describes the methods for measuring the real-ear attenuation of hearing protectors. Method A, trained-subject fit, was intended to describe the capabilities of the devices fitted by thoroughly trained users while Method B, inexperienced-subject fit, was intended to approximate the protection that can be attained by groups of informed users in workplace hearing conservation programs. Inexperienced subjects are no longer considered "naïve" according to ANSI S12.6 after 12 or more sessions measuring the attenuation of earplugs or semi-insert devices. However, an inexperienced subject that has received high quality video instructions may no longer be considered "naïve" or "inexperienced" even after just one session. AFRL conducted an ANSI S12.6-2008 Method B study to determine what effect, if any, high quality instructions had on the performance of naïve or inexperienced subjects and the number of trials where the subject could still be considered naïve or inexperienced. This experiment used ten subjects who completed three ANSI S12.6

measurements using the A-B-A training order and another 10 subjects completed the study using the B-A-B training order (A = high quality video instructions, B = short "earplug pillow-pack" written instructions). The attenuation results will be discussed and the implications for ANSI S12.6.

9:50

2aNSa2. Evaluation of variability in real-ear attenuation testing using a unique database—35 years of data from a single laboratory. Elliott H. Berger and Ronald W. Kieper (Personal Safety Div., 3M, 7911 Zionsville Rd., Indianapolis, IN 46268, elliott.berger@mmm.com)

The gold standard in measuring hearing protector attenuation since the late 1950s has been real-ear attenuation at threshold (REAT). Though well understood and standardized both in the U. S. (ANSI S3.19-1974 and ANSI S12.6-2008) and internationally (ISO 4869-1:1990), and known to provide valid and reliable estimates of protection for the test panel being evaluated, an area that is not clearly defined is the variability of the test measurements within a given laboratory. The test standards do provide estimates of uncertainty, both within and between laboratories, based on limited test data and interlaboratory studies, but thus far no published within-laboratory data over numerous tests and years have been available to provide empirical support for variability statements. This paper provides information from a one-of-a-kind database from a single laboratory that has conducted nearly 2500 studies over a period of 35 years in a single facility, managed by the same director (the lead author). Repeat test data on a controlled set of samples of a foam earplug, a premolded earplug, and two different earmuffs, with one of the data sets comprising 25 repeat tests over that 35-year period, will be used to demonstrate the inherent variability of this type of human-subject testing.

10:10

2aNSa3. Sound field uncertainty budget for real-ear attenuation at threshold measurement per ANSI S12.6 standards. Jeremie Voix and Céline Lapotre (École de technologie supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca)

In many national and international standards, the attenuation of Hearing Protection Devices is rated according to a psychophysical method called Real-Ear Attenuation at Threshold (REAT), which averages on a group of test-subjects the difference between the open and occluded auditory thresholds. In ANSI S12.6 standard, these REAT tests are conducted in a diffuse sound field in which sound uniformity and directionality are assessed by two objective microphone measurements. While the ANSI S12.6 standard defines these two criteria, it does not link the microphone measurements to the actual variation of sound pressure level at the eardrum that may originate from natural head movements during testing. This presentation examines this issue with detailed measurements conducted in an ANSI S12.6-compliant audiometric booth using an Artificial Test Fixture (ATF). The sound pressure level variations were recorded for movements of the ATF along the three main spatial axes and two rotation planes. From these measured variations and different head movements hypothetical scenarios, various sound field uncertainty budgets were computed. These findings will be discussed in order to eventually include them for uncertainty budget in a revised version of the ANSI S12.6 standard.

10:30

2aNSa4. Estimating effective noise dose when using hearing protection: Differences between ANSI S12.68 calculations and the auditory response measured with temporary threshold shifts. Hilary L. Gallagher, Richard L. McKinley (Battlespace Acoust., Air Force Res. Lab., AFRL/711HPW/RHCB, 2610 Seventh St, Wright-Patterson AFB, OH 45433-7901, hilary.gallagher.1@us.af.mil), Elizabeth A. McKenna (Ball Aerosp. and Technologies, Air Force Res. Lab., Wright-Patterson AFB, OH), and Mellisa A. Theis (ORISE, Air Force Res. Lab., Wright-Patterson AFB, OH)

ANSI S12.6 describes the methods for measuring the real-ear attenuation at threshold of hearing protectors. ANSI S12.68 describes the methods of estimating the effective A-weighted sound pressure levels when hearing protectors are worn. In theory, the auditory response, as measured by temporary threshold shifts (TTS), to an unoccluded ear noise exposure and an equivalent occluded ear noise exposure should produce similar behavioral results. In a series of studies conducted at the Air Force Research Laboratory, human subjects were exposed to continuous noise with and without hearing protection. Ambient noise levels during the occluded ear exposures were determined using ANSI S12.6 and ANSI S12.68. These equivalent noise exposures as determined by the ANSI S12.68 "gold standard" octave-band method produced significantly different auditory responses as measured with TTS. The methods and results from this study will be presented.

Contributed Papers

10:50

2aNSa5. Fit-testing, training, and timing—How long does it take to fit-test hearing protectors? Taichi Murata (Environ. Health Sci., Univ. of Michigan, School of Public Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226, ygo7@cdc.gov), Christa L. Themann, David C. Byrne, and William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH)

Hearing protector fit-testing is a Best Practice for hearing loss prevention programs and is gaining acceptance among US employers. Fit-testing quantifies hearing protector attenuation achieved by individual workers and ensures that workers properly fit and receive adequate protection from their protectors. Employers may be reluctant to conduct fit-testing because of expenses associated with worker time away from the job, personnel to administer the testing, and acquisition of a fit-test system. During field and laboratory studies conducted by the National Institute for Occupational Safety and Health (NIOSH), timing data for the fit-test process with the NIOSH HPD Well-Fit^{TM} system were analyzed. For workers completely naïve to fit-testing, the tests were completed within 15–20 minutes. Unoccluded test times were less than 4 minutes and occluded tests required less than 3 minutes. A significant learning effect was seen for the psychoacoustic method of adjustment used by HPD Well-Fit, explaining the shorter test times as subjects progressed through the unoccluded and occluded conditions. Most of the workers required about 5 minutes of training time. Test times and attenuations were tester-dependent, indicating the need to provide training to staff administering fit-tests in the workplace.

11:05

2aNSa6. Intra-subject fit variability using field microphone-in-real-ear attenuation measurement for foam, pre-molded and custom molded earplugs. Jeremie Voix (École de technologie supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca), Cecile Le Cocq (École de technologie supérieure, Université du Québec, Montreal, QC, Canada), and Elliott H. Berger (E•A•RCAL Lab, 3M Personal Safety Div., Indianapolis, IN)

In recent years, the arrival of several field attenuation estimation systems (FAES) on the industrial marketplace have enabled better assessment of hearing protection in real-life noise environments. FAES measure the individual attenuation of a given hearing protection device (HPD) as fitted by the end-user, but FAES enable predictions based only on measurements taken over a few minutes and do not account for what may occur later in the field over months or years as the earplug may be fitted slightly differently over time. This paper will use the field microphone-in-real-ear (F-MIRE) measurement technique to study in the laboratory how consistently a subject can fit and refit an HPD. A new metric, the intra-subject fit variability, will be introduced and quantified for three different earplugs (roll-down foam, premolded and custom molded), as fitted by two types of test subjects (experienced and inexperienced). This paper will present the experimental process used and statistical calculations performed to quantify intra-subject fit variability. As well, data collected from two different laboratories will be contrasted and reviewed as to the impact of trained versus untrained test subjects.

11:20

2aNSa7. A new perceptive method to measure active insertion loss of active noise canceling headsets or hearing protectors by matching the timbre of two audio signals. Remi Poncot and Pierre Guiu (Parrot S.A. France, 15 rue de montreuil, Paris 75011, France, poncotremi@gmail.com)

Attenuation of passive hearing protectors is assessed either by the Real Ear Attenuation at Threshold subjective method or by objective Measurements In the Real Ear. For Active Noise Cancelling headsets both methods do not practically apply. Alternative subjective methods based on loudness balance and masked hearing threshold techniques were proposed. However, they led to unmatched results with objective measurements at low frequency, diverging in either direction. Additionally, they are relatively long as frequency points of interest are measured one after the other. This paper presents a novel subjective method based on timbre matching, which has the originality of involving other perceptive mechanisms than the previous ones did. The attenuation performance of ANC headsets is rated by the change in pressure level of eight harmonics when the active noise reduction functionality is switched on. All harmonics are played at once, and their levels are adjusted by the test subject until he perceives the same timbre both in passive and active modes. A test was carried out by a panel of people in diffuse noise field conditions to assess the performance of personal consumer headphones. Early results show that the method is as repeatable as MIRE and lead to close results.

TUESDAY MORNING, 28 OCTOBER 2014 INDIANA E, 8:15 A.M. TO 11:20 A.M.

Session 2aNSb

Noise and Structural Acoustics and Vibration: Launch Vehicle Acoustics I

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

Seiji Tsutsumi, Cochair
JEDI Center, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara 252-5210, Japan

Chair's Introduction—8:15

Invited Papers

8:20

2aNSb1. Inclusion of source extent and coherence in a finite-impedance ground reflection model with atmospheric turbulence. Kent L. Gee and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

Acoustic data collected in static rocket tests are typically influenced by ground reflections. Furthermore, the partial coherence of the ground interaction due to atmospheric turbulence can play a significant role for larger propagation distances. Because the rocket plume is an extended radiator whose directionality is the result of significant source correlation, assessment of the impact of ground reflections in the data must include these effects. In this paper, a finite impedance-ground, single-source interference approach [G. A. Daigle, J. Acoust. Soc. Am. 65, 45–49 (1979)] that incorporates both amplitude and phase variations due to turbulence is extended to distributions of correlated monopoles. The theory for obtaining the mean-square pressure from multiple correlated sources in the presence of atmospheric turbulence is described. The effects of source correlation and extent, ground effective flow resistivity, and turbulence parameters are examined in terms differences in relative sound pressure level across a realistic parameter space. Finally, the model prediction is compared favorably against data from horizontal firings of large solid rocket motors. [Work supported by NASA MSFC and Blue Ridge Research and Consulting, LLC.]

8:40

2aNSb2. Estimation of acoustic loads on a launch vehicle fairing. Mir Md M. Morshed (Dept. of Mech. Eng., Jubail Univ. College, Jubail Industrial City, Jubail 10074, Saudi Arabia, morshedm@ucj.edu.sa), Colin H. Hansen, and Anthony C. Zander (School of Mech. Eng., The Univ. of Adelaide, Adelaide, SA, Australia)

During the launch of space vehicles, there is a large external excitation generated by acoustic and structural vibration. This is due to acoustic pressure fluctuations on the vehicle fairing caused by the engine exhaust gases. This external excitation drives the fairing structure and produces large acoustic pressure fluctuations inside the fairing cavity. The acoustic pressure fluctuations not only produce high noise levels inside the cavity but also cause damage such as structural fatigue, and damage to, or destruction of, the payload inside the fairing. This is an important problem because one trend of the aerospace industry is to use composite materials for the construction of launch vehicle fairings, resulted in large-scale weight reductions of launch vehicles, but increased the noise transmission inside the fairing. This work investigates the nature of the external acoustic pressure distribution on a representative small launch vehicle fairing during liftoff. The acoustic pressure acting on a representative small launch vehicle fairing was estimated from the complex acoustic field generated by the rocket exhaust during liftoff using a non-unique source allocation technique which considered acoustic sources along the rocket engine exhaust flow. Numerical and analytical results for the acoustic loads on the fairing agree well.

9:00

2aNSb3. Prediction of acoustic environments from horizontal rocket firings. Clothilde Giacomoni and Janice Houston (NASA/MSFC, NASA Marshall Space Flight Ctr., Bldg 4203, Cube 3128, Msfc, AL 35812, clothilde.b.giacomoni@nasa.gov)

In recent years, advances in research and engineering have led to more powerful launch vehicles which yield acoustic environments potentially destructive to the vehicle or surrounding structures. Therefore, it has become increasingly important to be able to predict the acoustic environments created by these vehicles in order to avoid structural and/or component failure. The current industry standard technique for predicting launch-induced acoustic environments was developed by Eldred in the early 1970s. Recent work has shown Eldred's technique to be inaccurate for current state-of-the-art launch vehicles. Due to the high cost of full-scale and even sub-scale rocket experiments, very little rocket noise data is available. Much of the work thought to be applicable to rocket noise has been done with heated jets. A model to predict the acoustic environment due to a launch vehicle in the far-field was created. This was done using five sets of horizontally fired rocket data, obtained between 2008 and 2012. Through scaling analysis, it is shown that liquid and solid rocket motors exhibit similar spectra at similar amplitudes. This model is accurate for these five data sets within 5 dB of the measured data.

9:20

2aNSb4. Acoustics research of propulsion systems. Ximing Gao (NASA Marshall Space Flight Ctr., Atlanta, Georgia) and Janice Houston (NASA Marshall Space Flight Ctr., 650 S. 43rd St., Boulder, Colorado 80305, janice.d.houston@nasa.gov)

The liftoff phase induces high acoustic loading over a broad frequency range for a launch vehicle. These external acoustic environments are used in the prediction of the internal vibration responses of the vehicle and components. Present liftoff vehicle acoustic environment prediction methods utilize stationary data from previously conducted hold-down tests to generate 1/3 octave band Sound Pressure Level (SPL) spectra. In an effort to update the accuracy and quality of liftoff acoustic loading predictions, non-stationary flight data from the Ares I-X were processed in PC-Signal in two flight phases: simulated hold-down and liftoff. In conjunction, the Prediction of Acoustic Vehicle Environments (PAVE) program was developed in MATLAB to allow for efficient predictions of sound pressure levels (SPLs) as a function of station number along the vehicle using semi-empirical methods. This consisted of generating the Dimensionless Spectrum Function (DSF) and Dimensionless Source Location (DSL) curves from the Ares I-X flight data. These are then used in the MATLAB program to generate the 1/3 octave band SPL spectra. Concluding results show major differences in SPLs between the hold-down test data and the processed Ares I-X flight data making the Ares I-X flight data more practical for future vehicle acoustic environment predictions.

9:40

2aNSb5. Acoustics associated with liquid rocket propulsion testing. Daniel C. Allgood (NASA SSC, Bldg. 3225, Stennis Space Ctr., MS 39529, Daniel.C.Allgood@nasa.gov)

Ground testing of liquid rocket engines is a necessary step towards building reliable launch vehicles. NASA Stennis Space Center has a long history of performing both developmental and certification testing of liquid propulsion systems. During these test programs, the propulsion test article, test stand infrastructure and the surrounding community can all be exposed to significant levels of acoustic energy for extended periods of time. In order to ensure the safety of both personnel and equipment, predictions of these acoustic environments are conducted on a routine basis. This presentation will provide an overview of some recent examples in which acoustic analysis has been performed. Validation of these predictions will be shown by comparing the predictions to acoustic data acquired during small- and full-scale engine hot-fire testing. Applications of semi-empirical and advanced computational techniques will be reviewed for both sea-level and altitude test facilities.

10:00–10:20 Break

10:20

2aNSb6. Post-flight acoustic analysis of Epsilon launch vehicle at lift-off. Seiji Tsutsumi (JAXA's Eng. Digital Innovation Ctr., JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Kyoichi Ui (Space Transportation Mission Directorate, JAXA, Tsukuba, Japan), Tatsuya Ishii (Inst. of Aeronautical Technol., JAXA, Chofu, Japan), Shinichiro Tokudome (Inst. of Space and Aeronautical Sci., JAXA, Sagamihara, Japan), and Kei Wada (Tokyo Office, Sci. Service Inc., Chuuou-ku, Japan)

Acoustic level both inside and outside the fairing is measured at the first Epsilon Launch Vehicle (Epsilon-1). The obtained data shows time-varying fluctuation due to the ascent of the vehicle. Equivalent stationary duration for such non-stationary flight data is determined based on the procedure described in NASA HDBK-7005. The launch pad used by the former M-V launcher is modified for the Epsilon based on the Computational Fluid Dynamics (CFD) and 1/42-scale model tests. Although the launch pad is compact and any water injection system is not installed, 10 dB reduction in overall sound pressure level (OASPL) is achieved due to the modification for the Epsilon, comparing with the M-V. Acoustic level inside the fairing satisfies the design requirement. Acoustic design of the launch pad developed here is revealed to be effective. Prediction of the acoustics level based on the Computational Fluid Dynamics (CFD) and subscale testing is also investigated by comparing with the flight measurement.

10:40

2aNSb7. Jet noise-based diagnosis of combustion instability in solid rocket motors. Hunki Lee, Taeyoung Park, Won-Suk Ohm (Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ohm@yonsei.ac.kr), and Dohyung Lee (Agency for Defense Development, Daejeon, South Korea)

Diagnosis of combustion instability in a solid rocket motor usually involves *in-situ* measurements of pressure in the combustor, a harsh environment that poses challenges in instrumentation and measurement. This paper explores the possibility of remote diagnosis of combustion instability based on far-field measurements of rocket jet noise. Because of the large pressure oscillations associated with combustion instability, the wave process in the combustor has many characteristic features of nonlinear acoustics such as shocks and limit cycles. Thus the remote detection and characterization of instability can be performed by listening for the tell-tale signs of the combustor nonlinear acoustics, buried in the jet noise. Of particular interest is the choice of nonlinear acoustic measure (e.g., among skewness, bispectra, and Howell-Morfey Q/S) that best brings out the acoustic signature of instability from the jet noise data. Efficacy of each measure is judged against the static test data of two tactical motors (one stable, the other unstable).

11:00

2aNSb8. Some recent experimental results concerning turbulent coanda wall jets. Caroline P. Lubert (Mathematics & Statistics, James Madison Univ., 301 Dixie Ave., Harrisonburg, VA 22801, lubertcp@jmu.edu)

The Coanda effect is the tendency of a stream of fluid to stay attached to a convex surface, rather than follow a straight line in its original direction. As a result, in such jets mixing takes place between the jet and the ambient air as soon as the jet issues from its exit nozzle, causing air to be entrained. This air-jet mixture adheres to the nearby surface. Whilst devices employing the Coanda effect usually offer substantial flow deflection, and enhanced turbulence levels and entrainment compared with conventional jet flows, these prospective advantages are generally accompanied by significant disadvantages including a considerable increase in associated noise levels and jet breakaway. Generally, the reasons for these issues are not well understood and thus the full potential offered by the Coanda effect is yet to be realized. The development of a model for predicting the noise emitted by three-dimensional flows over Coanda surfaces would suggest ways in which the noise could be reduced or attenuated. In this paper, the results of recent experiments on a 3-D turbulent Coanda wall jet are presented. They include the relationship of SPL, shock cell distribution and breakaway to various flow parameters, and predictions of the jet boundary.

Session 2aPA

Physical Acoustics: Outdoor Sound Propagation

Kai Ming Li, Cochair
Mechanical Engineering, Purdue University, 140 South Martin Jischke, West Lafayette, IN 47907-2031

Shahram Taherzadeh, Cochair
Engineering & Innovation, The Open University, Walton Hall, Milton Keynes MK7 6AA, United Kingdom

Contributed Papers

8:30

2aPA1. On the inversion of sound fields above a locally reacting ground for direct impedance deduction. Kai Ming Li and Bao N. Tong (Mech. Eng., Purdue Univ., 177 South Russel St., West Lafayette, IN 47907-2099, mmkmli@purdue.edu)

A complex root-finding algorithm is typically used to deduce the acoustic impedance of a locally reacting ground by inverting the measured sound fields. However, there is an issue of uniquely determining the impedance from a measurement of an acoustic transfer function. The boundary loss factor F, which is a complex function, is the source of this ambiguity. It is associated with the spherical wave reflection coefficient Q for the reflected sound field. These two functions are dependent on a complex parameter known as the numerical distance w. The inversion of F leading to the multiple solutions of w can be identified as the root cause of the problem. To resolve this ambiguity, the zeroes and saddle points of F are determined for a given source/receiver geometry and a known acoustic impedance. They are used to establish the basins containing all plausible solutions. The topography of Q is further examined in the complex w-plane. A method for identifying the family of solutions and selecting the physically meaningful branch is proposed. Validation is provided by using numerical simulations as well as the experimentally data. The error and uncertainties in the deduced impedance are quantified.

8:45

2aPA2. An improved method for direct impedance deduction of a locally reacting ground. Bao N. Tong and Kai Ming Li (Mech. Eng., Purdue Univ., 177 South Russel St., West Lafayette, IN 47907-2099, bntong@purdue.edu)

An accurate deduction of the acoustic impedance of a locally reacting ground depends on a precise measurement of sound fields at short-ranges. However, measurement uncertainties exists in both the magnitude and the phase of the acoustic transfer function. By using the standard method, accurate determination of the acoustic impedance can be difficult when the measured phases become unreliable in many outdoor conditions. An improved technique, which relies only on the magnitude information, has been developed. A minimum of two measurements at two source/receiver configurations are needed to determine the acoustic impedance. Even in the absence of measurement uncertainties, a more careful analysis suggests that a third independent measurement is often needed to give an accurate solution. When experimental errors are inevitably introduced, a selection of optimal geometry becomes necessary to reduce the sensitivity of the deduced impedance to small variations in the data. A graphical method is provided which offers greater insight into the deduction of impedance and a downhill simplex algorithm has been developed to automate the procedure. Physical constraints are applied to limit the search region and to eliminate the rogue solutions. Several case studies using indoor and outdoor data are presented to validate the proposed technique.

9:00

2aPA3. Wavelet-like models for random media in wave propagation simulations. D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and 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), and Sergey N. Vecherin (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Dev. Ctr., Hanover, NH)

Simulations of wave propagation and scattering in random media are often performed by synthesizing the media from Fourier modes, in which the phases are randomized and the amplitudes tailored to provide a prescribed spectrum. Although this approach is computationally efficient, it cannot capture organization and intermittency in random media, which impacts higher-order statistical properties. As an alternative, we formulate a cascade model involving distributions of wavelet-like objects (quasi-wavelets or QWs). The QW model is constructed in a self-similar fashion, with the sizes, amplitudes, and numbers of offspring objects occurring at a constant ratios between generations. The objects are randomly distributed in space according to a Poisson process. The QW model is formulated in static (time-invariant), steady-state, and non-steady versions. Many diverse natural and man-made environments can be synthesized, including turbulence, porous media, rock distributions, urban buildings, and vegetation. The synthesized media can then be used in simulations of wave propagation and scattering.

9:15

2aPA4. Space-time correlation of acoustic signals in a turbulent atmosphere. Vladimir E. Ostashev, D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu), Sandra Collier (U.S. Army Res. Lab., Adelphi, MD), and Sylvain Cheinet (French-German Res. Inst. of Saint-Louis, Saint-Louis, France)

Scattering by atmospheric turbulence diminishes the correlation, in both space and time, of acoustic signals. This decorrelation subsequently impacts beamforming, averaging, and other techniques for enhancing signal-to-noise ratio. Space-time correlation can be measured directly with a phased microphone array. In this paper, a general theory for the space-time correlation function is presented. The atmospheric turbulence is modeled using the von Karman spatial spectra of temperature and wind velocity fluctuations and locally frozen turbulence (i.e., the Taylor's frozen turbulence hypothesis with convection velocity fluctuations). The theory developed is employed to calculate and analyze the spatial and temporal correlation of acoustic signals for typical regimes of an unstable atmospheric boundary layer, such as mostly cloudy or sunny conditions with light, moderate, or strong wind. The results obtained are compared with available experimental data.

9:30

2aPA5. Characterization of wind noise by the boundary layer meteorology. Gregory W. Lyons and Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of MS, 1 Coliseum Dr., University, MS 38677, gwlyons@go.olemiss.edu)

The fluctuations in pressure generated by turbulent motions of the atmospheric boundary layer are a principal noise source in outdoor acoustic measurements. The mechanics of wind noise involve not only stagnation pressure fluctuations at the sensor, but also shearing and self-interaction of turbulence throughout the flow, particularly at low frequencies. The contributions of these mechanisms can be described by the boundary-layer meteorology. An experiment was conducted at the National Wind Institute's 200-meter meteorological tower, located outside Lubbock, Texas in the Llano Estacado region. For two days, a 44-element 400-meter diameter array of unscreened NCPA-UMX infrasound sensors recorded wind noise continuously, while the tower and a Doppler SODAR measured vertical profiles of the boundary layer. Analysis of the fluctuating pressure with the meteorological data shows that the statistical structure of wind noise depends on both mean velocity distribution and buoyant stability. The root-mean-square pressure exhibits distinct scalings for stable and unstable stratification. Normalization of the pressure power spectral density depends on the outer scales. In stable conditions, the kurtosis of the wind noise increases with Reynolds number. Measures of noise intermittency are explored with respect to the meteorology.

9:45

2aPA6. Statistical moments for wideband acoustic signal propagation through a turbulent atmosphere. Jericho E. Cain (US Army Res. Lab., 1200 East West Hwy, Apt. 422, Silver Spring, MD 20910, jericho.cain@gmail.com), Sandra L. Collier (US Army Res. Lab., Adelphi, MD), Vladimir E. Ostashev, and David K. Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Developing methods for managing noise propagation, sound localization, sound classification, and for designing novel acoustic remote sensing methods of the atmosphere requires a detailed understanding of the impact that atmospheric turbulence has on acoustic propagation. In particular, knowledge of the statistical moments of the sound field is needed. The first statistical moment corresponds to the coherent part of the sound field and it is need in beamforming applications. The second moment enables analysis of the mean intensity of a pulse in a turbulent atmosphere. Numerical solutions to a set of recently derived closed form equations for the first and second order statistical moments of a wideband acoustic signal propagating in a turbulent atmosphere with spatial fluctuations in the wind and temperature fields are presented for typical regimes of the atmospheric boundary layer.

10:00–10:15 Break

10:15

2aPA7. Analysis of wind noise reduction by semi-porous fabric domes. Sandra L. Collier (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil), Richard Raspet (National Ctr. for Physical Acoust., Univ. of MS, University, MS), John M. Noble, W. C. Kirkpatrick Alberts (U.S. Army Res. Lab., Adelphi, MD), and Jeremy Webster (National Ctr. for Physical Acoust., Univ. of MS, University, MS)

For low frequency acoustics, the wind noise contributions due to turbulence may be divided into turbulence–sensor, turbulence–turbulence, and turbulence–mean shear interactions. Here, we investigate the use of a semi-porous fabric dome for wind noise reduction in the infrasound region. Comparisons are made between experimental data and theoretical predictions from a wind noise model [Raspet, Webster, and Naderyan, J. Acoust. Soc. Am. **135**, 2381 (2014)] that accounts for contributions from the three turbulence interactions.

10:30

2aPA8. An investigation of wind-induced and acoustic-induced ground motions. Vahid Naderyan, Craig J. Hickey, and Richard Raspet (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, NCPA, 1 Coliseum Dr.,, University, MS 38677, vnaderya@go.olemiss.edu)

Wind noise at low frequency is a problem in seismic surveys, which reduces seismic image clarity. In order to find a solution for this problem, we investigated the driving pressure perturbations on the ground surface associated with wind-induced ground motions. The ground surface pressure and shear stress at the air–ground interface were used to predict the displacement amplitudes of the horizontal and vertical ground motions as a function of depth. The measurements were acquired at a site having a flat terrain and low seismic ambient noise under windy conditions. Multiple tri-axial geophones were deployed at different depths to study the induced ground velocity as a function of depth. The measurements show that the wind excites horizontal components more than vertical component on the above ground geophone due to direct interaction with the geophone. For geophones buried flush with the ground surface and at various depths below the ground, the vertical components of the velocity are greater than the horizontal components. There is a very small decrease in velocity with depth. The results are compared to acoustic-ground coupling case. [This work is supported by USDA under award 58-6408-1-608.]

10:45

2aPA9. Using an electro-magnetic analog to study acoustic scattering in a forest. Michelle E. Swearingen (US Army ERDC, Construction Eng. Res. Lab., P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil) and Donald G. Albert (US Army ERDC, Hanover, NH)

Using scale models can be a convenient method for investigating multiple scattering in complex environments, such as a forest. However, the increased attenuation with increasing frequency limits the propagation distances available for such models. An electromagnetic analog is an alternative way to study multiple scattering from rigid objects, such as tree trunks. This analog does not suffer from the intrinsic attenuation and allows for investigation of a larger effective area. In this presentation, the results from a 1:50 scale electromagnetic analog are compared to full-scale data collected in a forest. Further tests investigate propagation along multiple paths through a random configuration of aluminum cylinders representing trees. Special considerations and anticipated range of applicability of this analog method are discussed.

11:00

2aPA10. Modeling of sound scattering by an obstacle located below a hardbacked rigid porous medium. Yiming Wang and Kai Ming Li (Mech. Eng., Purdue Univ., 177 South Russel St., West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

The boundary integral equation (BIE) formulation takes advantage of the well-known Green's function for the sound fields above a plane interface. It can then lead to a simplified numerical solution known as the boundary element method (BEM) that enables an accurate computation of sound fields above the plane interface with the presence of obstacles of complex shapes. The current study is motivated by the need to explore the acoustical characteristics of a layer of sound absorption materials embedded with equally spaced rigid inserts. In principle, this problem may be solved by a standard finite element program but it is found more efficient to use the BIE approach by discretizing only the boundary surfaces of the obstacles within the medium. The formulation is facilitated by using accurate Green's functions for computing the sound fields above and within a layer of rigid porous medium. This paper reports a preliminary study to model the scattering of sound by an obstacle placed within the layered rigid porous medium. The two-dimensional Green's functions will be derived and used for the development of a BEM model for computing the sound field above and within the rigid porous medium due to the presence of an arbitrarily shaped obstacle.

11:15

2aPA11. Analysis of the Green's function for a duct and cavity using geometric image sources. Ambika Bhatta, Charles Thompson, and Kavitha Chandra (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu)

The presented work investigates the solution for pressure response of a point source in a two dimensional waveguide. The methodology is based on the one dimensional analytical and numerical solution of a finite channel response between two semi-infinite planes. The branch integrals representing the reflection coefficient is implemented to evaluate the pressure amplitude of the boundary effect. The approach addresses the validation of application of geometric image sources for finite boundaries. Consequently, the 3D extension of the problem for a closed cavity is also investigated.

TUESDAY MORNING, 28 OCTOBER 2014 MARRIOTT 1/2, 8:00 A.M. TO 10:00 A.M.

Session 2aSAa

Structural Acoustics and Vibration and Noise: Computational Methods 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

Matthew Kamrath, Cochair
Acoustics, Pennsylvania State University, 717 Shady Ridge Road, Hutchinson, MN 55350

Invited Papers

8:00

2aSAa1. A radical technology for modeling target scattering. David Burnett (Naval Surface Warfare Ctr., Code CD10, 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

NSWC PCD has developed a high-fidelity 3-D finite-element (FE) modeling system that computes acoustic color templates (target strength vs. frequency and aspect angle) of single or multiple realistic objects (e.g., target + clutter) in littoral environments. High-fidelity means that 3-D physics is used in all solids and fluids, including even thin shells, so that solutions include not only all propagating waves but also all evanescent waves, the latter critically affecting the former. Although novel modeling techniques have accelerated the code by several orders of magnitude, NSWC PCD is now implementing a radically different FE technology, e.g., one thin-shell element spanning 90° of a cylindrical shell. It preserves all the 3-D physics but promises to accelerate the code another two to three orders of magnitude. The talk will briefly review the existing system and then describe the new technology.

8:20

2aSAa2. Faster frequency sweep methods for structural vibration and acoustics analyses. Kuangcheng Wu (Ship Survivability, Newport News ShipBldg., 202 Schembri Dr., Yorktown, VA 23693, kc.wu@hii-nns.com) and Vincent Nguyen (Ship Survivability, Newport News ShipBldg., Newport News, VA)

The design of large, complex structures typically requires knowledge of the mode shape and forced response near major resonances to ensure deflection, vibration, and the resulting stress are kept below acceptable levels, and to guide design changes where necessary. Finite element analysis (FEA) is commonly used to predict Frequency Response Functions (FRF) of the structure. However, as the complexity and detail of the structure grows, the system matrices, and the computational resources needed to solve them, get large. Furthermore, the need to use small frequency steps to accurately capture the resonant response peaks can drive up the number of FRF calculations required. Thus, the FRF calculation can be computationally expensive for large structural systems. Several approaches have been proposed that can significantly accelerate the overall process by approximating the frequency dependent response. Approximation approaches based on Krylov Galerkin Projection (KGP) and Pade calculate the forced response at only a few frequencies, then use the response and its derivatives to reconstruct the FRF in-between the selected direct calculation points. This paper first validates the two approaches with analytic solutions for a simply supported plate, and then benchmarks several numerical examples to demonstrate the accuracy and efficiency of the new approximate methods.

8:40

2aSAa3. Waves in continua with extreme microstructures. Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu)

The effective properties of a material may be generally defined as those that describe the limiting case where the wavelength of propagation is infinite compared to the characteristic scale of the microstructure. Generally, the limit of vanishingly small microstructural scale in a heterogeneous elastic medium results in an effective homogeneous medium that is again elastic. We show that for materials with extreme microstructures, the limiting effective medium can be quite exotic, including polar materials, or multiphase continuum. These continuum models naturally give rise to unusual effective properties including negative or anisotropic mass. Though unusual, these properties have straightforward interpretations in terms of the laws of classical mechanics. Finally, we discuss wave propagation in these structures and find dispersion curves with multiple branches.

Contributed Papers

9:00

2aSAa4. A comparison of perfectly matched layers and infinite elements for exterior Helmholtz problems. Gregory Bunting (Computational Solid Mech. and Structural Dynam., Sandia National Labs., 709 Palomas Dr. NE, Albuquerque, NM 87108, bunting.gregory@gmail.com), Arun Prakash (Lyles School of Civil Eng., Purdue Univ., West Lafayette, IN), and Timothy Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., West Lafyette, IN)

Perfectly matched layers and infinite elements are commonly used for finite element simulations of acoustic waves on unbounded domains. Both involve a volumetric discretization around the periphery of an acoustic mesh, which itself surrounds a structure or domain of interest. Infinite elements have been a popular choice for these problems since the 1970s. Perfectly matched layers are a more recent technology that is gaining popularity due to ease of implementation and effectiveness as an absorbing boundary condition. In this study, we present massively parallel implementations of these two techniques, and compare their performance on a set of representative structural–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 effects of PML parameters, exterior boundary conditions, and quadrature rules on the accuracy of the solution. [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.]

9:15

2aSAa5. Improved model for coupled structural–acoustic modes of tires. Rui Cao, Nicholas Sakamoto, and J. S. Bolton (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 S. Russell St., West Lafayette, IN 47907-2099, cao101@purdue.edu)

Experimental measurements of tire tread band vibration have provided direct evidence that higher order structural-acoustic modes exist in tires, not just the well-know fundamental mode. These modes display both circumferential and radial pressure variations. The theory governing these modes has thus been investigated. A brief recapitulation of the previously-presented coupled tire-acoustical model based on a tensioned membrane approach will be given, and then an improved tire-acoustical model with a ring-like shape will be introduced. In the latter model, the effects of flexural and circumferential stiffness are considered, as is the role of curvature in coupling the various wave types. This improved model accounts for propagating in-plane vibration in addition to the essentially structure-borne flexural wave and the essentially airborne longitudinal wave accounted for in the previous model. The longitudinal structure-borne wave "cuts on" at the tire's circumferential ring frequency. Explicit solutions for the structural and acoustical modes will be given in the form of dispersion relations. The latter results will be compared with measured dispersion relations, and the features associated primarily with the higher order acoustic modes will be highlighted. Finally, the effect of tire rotational speed on the natural frequencies of these various modes types will also be discussed.

9:30

2aSAa6. Simulating sound absorption in porous material with the lattice Boltzmann method. Andrey R. da Silva (Ctr. for Mobility Eng., Federal Univ. of Santa Catarina, Rua Monsenhor Topp, 173, Florianópolis, Santa Catarina 88020-500, Brazil, andrey.rs@ufsc.br), Paulo Mareze, and Eric Brandão (Structure and Civil Eng., Federal Univ. of Santa Maria, Santa Maria, RS, Brazil)

The development of porous materials that are able to absorb sound in specific frequency bands has been an important challenge in the acoustic research. Thus, the development new numerical techniques that allow one to correctly capture the mechanisms of sound absorption can be seen as an important step to developing new materials. In this work, the lattice Boltzmann method is used to predict the sound absorption coefficient in porous material with straight porous structure. Six configurations of porous material were investigated, involving different thickness and porosity values. A very good agreement was found between the numerical results and those obtained by the analytical model provided in the literature. The results suggest that the lattice Boltzmann model can be a powerful alternative to simulating viscous sound absorption, particularly due to its reduced computational effort when compared to traditional numerical methods.

9:45

2aSAa7. Energy flow models for the out-of-plane vibration of horizontally curved beams. Hyun-Gwon Kil (Dept. of Mech. Eng., Univ. of Suwon, 17, Wauan-gil, Bongdam-eup, Hwaseong-si, Gyeonggi-do 445-743, South Korea, hgkil@suwon.ac.kr), Seonghoon Seo (Noise & Vib. CAE Team, Hyundai Motor Co., Hwaseong-si, Gyeonggi-do, South Korea), Suk-Yoon Hong (Dept. of Naval Architecture and Ocean Eng., Seoul National Univ., Seoul, South Korea), and Chan Lee (Dept. of Mech. Eng., Univ. of Suwon, Hwaseong-si, Gyeonggi-do, South Korea)

The purpose of this work is to develop energy flow models to predict the out-of-plane vibration of horizontally curved beams in the mid- and high-frequency range. The dispersion relations of waves are approximately separated into relations to the propagation of flexural waves and torsional waves generating the out-of-plane vibration of the horizontally curved beams with Kirchhoff-Love hypotheses. The energy flow models are based on the energy governing equations for the flexural waves and the torsional waves propagating in the curved beams. Those equations are driven to predict the time- and locally space-averaged energy density and intensity in the curved beams. Total values for the energy density and the intensity as well as contributions of each type of waves on those values are predicted. A verification of the energy flow models for the out-of-plane vibration of the horizontally curved beams is performed by comparing the energy flow solutions for the energy density and the intensity with analytical solutions evaluated using the wave propagation approach. The comparison shows that the energy flow models can be effectively used to predict the out-of-plane vibration of the horizontally curved beams in the mid- and high-frequency range.

Session 2aSAb

Structural Acoustics and Vibration and Noise: Vehicle Interior Noise

Sean F. Wu, Chair
Mechanical Engineering, Wayne State University, 5050 Anthony Wayne Drive, College of Engineering Building, Rm. 2133, Detroit, MI 48202

Chair's Introduction—10:30

Invited Papers

10:35

2aSAb1. Structural–acoustic optimization of a pressurized, ribbed aircraft panel. 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)

A method to reduce the noise radiated by a ribbed, aircraft panel excited by turbulent boundary layer flow is presented. To compute the structural-acoustic response, a modal approach based on finite element/boundary element analysis was coupled to a turbulent boundary flow forcing function. A static pressure load was also applied to the panel to simulate cabin pressurization during flight. The radiated sound power was then minimized by optimizing the horizontal and vertical rib location and rib cross section using an evolutionary search algorithm. Nearly 10 dB of reduction was achieved by pushing the ribs to the edge of the panel, thus lowering the modal amplitudes excited by the forcing function. A static constraint was then included in the procedure using a low-frequency dynamic calculation to approximate the static response. The constraint limited the amount of reduction that was achieved by the optimizer.

11:00

2aSAb2. Extending interior near-field acoustic holography to visualize three-dimensional objective parameters of sound quality. Huancai Lu (Mech. Eng., Zhejiang Univ. of Technol., 3649 Glenwood Ave., Windsor, ON N9E 2Y6, Canada, huancailu@zjut.edu.cn)

It is essential to understand that the ultimate goal of interior noise control is to improve the sound quality inside the vehicle, rather than to suppress the sound pressure level. Therefore, the vehicle interior sound source localization and identification should be based on the contributions of sound sources to the subjective and/or objective parameters of sound quality at targeted points, such as driver's ear positions. This talk introduces the visualization of three-dimensional objective parameters of sound quality based on interior near-field acoustic holography (NAH). The methodology of mapping three-dimensional sound pressure distribution, which is reconstructed based on interior NAH, to three-dimensional loudness is presented. The mathematical model of loudness developed by ANSI standard is discussed. The numerical interior sound field, which is generated by vibrating enclosure with known boundary conditions, is employed to validate the methodology. In addition, the accuracy of reconstruction of loudness distribution is examined with ANSI standard and digital head. It is shown that the results of sound source localization based on three-dimensional loudness distribution are different from the ones based on interior NAH.

Contributed Paper

11:25

2aSAb3. A comparative analysis of the Chicago Transit Authority's Red Line railcars. Chris S. Nottoli (Riverbank Acoust. Labs., 1145 Walter, Lemont, IL 60439, cnottoli18@gmail.com)

A noise study was conducted on Chicago Transit Authority's Red Line railcars to assess the differences in interior sound pressure level between the 5000 series railcars and its predecessor, the 2400 series. The study took into account potential variability associated with a rider's location in the railcars, above ground, and subway segments (between stations), and surveyed the opinion of everyday Red Line riders as pertaining to perceived noise. The test data demonstrated a 3–6 dB noise reduction in ongoing CTA renovations between new rapid transit cars and their predecessors. Location on the train influenced Leq(A) measurements as reflections from adjacent railcars induced higher noise levels. The new railcars also proved effective in noise reduction throughout the subway segments as the averaged Leq(A) deviated 1 dB from above ground rail stations. Additionally, this study included an online survey that revealed a possible disconnect between traditional methods of objective noise measurement and subjective noise ratings.

Session 2aSC

Speech Communication: Speech Production and Articulation (Poster Session)

Sam Tilsen, Chair
Cornell University, 203 Morrill Hall, Ithaca, NY 14853

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

2aSC1. Tongue motion characteristics during vowel production in older children and adults. Jennell Vick, Michelle Foye (Psychol. Sci., Case Western Reserve Univ., 11635 Euclid Ave., Cleveland, OH 44106, jennell@case.edu), Nolan Schreiber, Greg Lee (Elec. Eng. Comput. Sci., Case Western Reserve Univ., Cleveland, OH), and Rebecca Mental (Psychol. Sci., Case Western Reserve Univ., Cleveland, OH)

This study examined tongue movements in consonant-vowel-consonant sequences drawn from real words in phrases as produced by 36 older children (three male and three female talkers at each age from 10 to 15 years) and 36 adults. Movements of four points on the tongue were tracked at 400 Hz using the Wave Electromagnetic Speech Research System (NDI, Waterloo, ON, CA). The four points were tongue tip (TT; 1 cm from tip on midline), tongue body (TB; 3 cm from tip on midline), tongue right (TR; 2 cm from tip on right lateral edge), and tongue left (TR; 2 cm from tip on left lateral edge). The phrases produced included the vowels /i/, /I/, /ae/, and /u/ in words (i.e., "see," sit," cat," and "zoo"). Movement measures included 3D distance, peak and average speed, and duration of vowel opening and closing strokes. The horizontal curvature of the tongue was calculated at the trajectory speed minimum associated with the vowel production using a least-squares quadratic fit of the TR, TB, and TL positional coordinates. Symmetry of TR and TL vertical position was also calculated. Within-group comparisons were made between vowels and between-group comparisons were made between children and adults.

2aSC2. Experimental evaluation of the constant tongue volume hypothesis. Zisis Iason Skordilis, Vikram Ramanarayanan (Signal Anal. and Interpretation Lab., Dept. of Elec. Eng., Univ. of Southern California, 3710 McClintock Ave., RTH 320, Los Angeles, CA 90089, skordili@usc.edu), Louis Goldstein (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA), and Shrikanth S. Narayanan (Signal Anal. and Interpretation Lab., Dept. of Elec. Eng., Univ. of Southern California, Los Angeles, CA)

The human tongue is considered to be a muscular hydrostat (Kier and Smith, 1985). As such, it is considered to be incompressible. This constant volume hypothesis has been incorporated in various mathematical models of the tongue, which attempt to provide insights into its dynamics (e.g., Levine *et al.*, 2005). However, to the best of our knowledge, this hypothesis has not been experimentally validated for the human tongue during actual speech production. In this work, we attempt an experimental evaluation of the constant tongue volume hypothesis. To this end, volumetric structural Magnetic Resonance Imaging (MRI) was used. A database consisting of 3D MRI images of subjects articulating continuants was considered. The subjects sustained contextualized vowels and fricatives (e.g., IY in "beet," F in "afa") for 8 seconds in order for the 3D geometry to be collected. To segment the tongue and estimate its volume, we explored watershed (Meyer and Beucher, 1990) and region growing (Adams and Bischof, 1994) techniques. Tongue volume was estimated for each lingual posture for each subject. Intra-subject tongue volume variation was examined to determine if there is sufficient statistical evidence for the validity of the constant volume hypothesis. [Work supported by NIH and a USC Viterbi Graduate School Ph.D. fellowship.]

2aSC3. A physical figure model of tongue muscles. Makoto J. Hirayama (Faculty of Information Sci. and Technol., Osaka Inst. of Technol., 1-79-1 Kitayama, Hirakata 573-0196, Japan, mako@is.oit.ac.jp)

To help understanding tongue shape and motions, a physical figure model of tongue muscles using viscoelastic material of urethane rubber gel were made by improving previous models. Compare to previous shape tongue models that had been made and presented, the new model is constructed from tongue body (consisting of Transversus linguae, Verticalis linguae, Longitudinalis linguae superior, and Longitudinalis linguae inferior), and individual extrinsic tongue muscles (consisting of Genioglossus anterior, Genio glossus posterior, Hyoglossus, Styloglossus, and Palatoglossus) parts. Therefore, each muscle's shape, starting and ending points, and relation to other muscles and organs inside mouth are more understandable than previous ones. As the model is made from viscoelastic material similar to human skin, reshaping and moving tongue are possible by pulling or pushing some parts of the tongue muscle by hand, that is, tongue shape and motion simulations by hand can be done. The proposed model is useful for speech science education or a future speaking robot using realistic speech mechanism.

2aSC4. Tongue width at rest versus tongue width during speech: A comparison of native and non-native speakers. Sunao Kanada and Ian Wilson (CLR Phonet. Lab, Univ. of Aizu, Tsuruga, Ikki machi, Aizuwkamatsu, Fukushima 965-8580, Japan, m5181137@u-aizu.ac.jp)

Most pronunciation researchers do not focus on the coronal view. However, it is also important to observe because the tongue is hydrostatic. We believe that some pronunciation differences between native speakers and second-language (L2) speakers could be due to differences in the coronal plane. Understanding these differences could be a key to L2 learning and modeling. It may be beneficial for pedagogical purposes and the results of this research may contribute to the improvement of pronunciation of L2 English speakers. An interesting way to look at native and L2 articulation differences is through the pre-speech posture and inter-speech posture (ISP—rest position between sentences). In this research, we compare native speakers to L2 speakers. We measure how different those postures are from the median position of the tongue during speech. We focus on movement of a side tongue marker in the coronal plane, and we normalize for speaker size. We found that the mean distance from pre-speech posture to speech posture is shorter for native English speakers (0.95 mm) than for non-native English speakers (1.62 mm). So, native speakers are more efficient in their pre-speech posture. Results will also be shown for distances from ISP to speech posture.

2aSC5. Intraglottal velocity and pressure measurements in a hemilarynx model. Liran Oren, Sid Khosla (Otolaryngol., Univ. of Cincinnati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu), and Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH)

Determining the mechanisms of self-sustained oscillation of the vocal folds requires characterization of intraglottal aerodynamics. Since most of the intraglottal aerodynamics forces cannot be measured in a tissue model of the larynx, most of the current understanding of vocal fold vibration mechanism is derived from mechanical, analytical, and computational models. In the current study, intraglottal pressure measurements are taken in a hemilarynx model and are compared with pressure values that are computed form simultaneous velocity measurements. The results show that significant negative pressure is formed near the superior aspect of the folds during closing, which is in agreement with previous measurements in a hemilarynx model. Intraglottal velocity measurements show that the flow near the superior aspect separates from the glottal wall during closing and may develop into a vortex, which further augments the magnitude of the negative pressure. The intraglottal pressure distributions are computed by solving the pressure Poisson equation using the velocity field measurements and show good agreement with the pressure measurements. The match between the pressure computations and the pressure measurements validates the technique, which was also used in previous study to estimate the intraglottal pressure distribution in a full larynx model.

2aSC6. Ultrasound study of diaphragm motion during tidal breathing and speaking. Steven M. Lulich, Marguerite Bonadies (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu), Meredith D. Lulich (Southern Indiana Physicians, Indiana Univ. Health, Bloomington, IN), and Robert H. Withnell (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Studies of speech breathing by Ladefoged and colleagues (in the 1950s and 1960s), and by Hixon and colleagues (in the 1970s, 1980s, and 1990s) have substantially contributed to our understanding of respiratory mechanics during speech. Even so, speech breathing is not well understood when contrasted with phonation, articulation, and acoustics. In particular, diaphragm involvement in speech breathing has previously been inferred from inductive plethysmography and EMG, but it has never been directly investigated. In this case study, we investigated diaphragm motion in a healthy adult male during tidal breathing and conversational speech using real-time 3D ultrasound. Calibrated inductive plethysmographic data were recorded simultaneously for comparison with previous studies and in order to relate lung volumes directly to diaphragm motion.

2aSC7. A gestural account of Mandarin tone sandhi. Hao Yi and Sam Tilsen (Dept. of Linguist, Cornell Univ., 315-7 Summerhill Ln., Ithaca, NY 14850, hy433@cornell.edu)

Recently tones have been analyzed as articulatory gestures, which may be coordinated with segmental gestures. Our data from electromagnetic articulometry (EMA) show that purported neutralized phonological contrast can nonetheless exhibit coordinative difference. We develop a model based on gestural coupling to account for observed patterns. Mandarin Third Tone Sandhi (e.g., Tone3 $\rightarrow$ T3S /_ Tone3) is perceptually neutralizing in that the sandhi output (T3S) shares great similarity with Tone2. Despite both tones having rising pitch contours, there exist subtle acoustic differences. However, the difference in underlying representation between T3S and Tone2 remains unclear. By presenting evidence from the alignment pattern between tones and segments, we show that the acoustic differences between Tone2 and T3S arises out of the difference in gestural organizations. The temporal lag between the initiation of the Vowel gesture and that of Tone gesture in T3S is shorter than that in Tone2. We further argue that underlying Tone3 is the source of incomplete neutralization between the Tone2 and T3S. That is, despite the surface similarity, T3S is stored in the mental lexicon as Tone3.

2aSC8. A real-time MRI investigation of anticipatory posturing in prepared responses. Sam Tilsen (Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, tilsen@cornell.edu), Pascal Spincemaille (Radiology, Cornell Weill Medical College, New York, NY), Bo Xu (Biomedical Eng., Cornell Univ., New York, NY), Peter Doerschuk (Biomedical Eng., Cornell Univ., Ithaca, NY), Wenming Luh (Human Ecology, Cornell Univ., Ithaca, NY), Robin Karlin, Hao Yi (Linguist, Cornell Univ., Ithaca, NY), and Yi Wang (Biomedical Eng., Cornell Univ., Ithaca, NY)

Speakers can anticipatorily configure their vocal tracts in order to facilitate the production of an upcoming vocal response. We find that this anticipatory articulation results in decoherence of articulatory movements that are otherwise coordinated; moreover, speakers differ in the strategies they employ for response anticipation. Real-time MRI images were acquired from eight native English speakers performing a consonant-vowel response task; the task was embedded in a 2×2 design, which manipulated preparation (whether speakers were informed of the target response prior to a go-signal) and postural constraint (whether the response was preceded by a prolonged vowel). Analyses of pre-response articulatory postures show that all speakers exhibited anticipatory posturing of the tongue root in unconstrained responses. Some exhibited interactions between preparation and constraint, such that anticipatory posturing was more extensive in prepared- vs. unprepared-unconstrained responses. Cross-speaker variation was also observed in anticipatory posturing of the velum: some speakers raised the velum in anticipation of non-nasal responses, while others failed to do so. The results show that models of speech production must be flexible enough to allow for gestures to be executed individually, and that speakers differ in the strategies they employ for response initiation.

2aSC9. An airflow examination of the Czech trills. Ekaterina Komova (East Asian Lang. and Cultures, Columbia Univ., New York, NY) and Phil Howson (The Univ. of Toronto, 644B-60 Harbord St., Toronto, ON M5S3L1, Canada, phil.howson@mail.utoronto.ca)

Previous studies have suggested that there is a difference between the Czech trills /r̝/ and /r/ with respect to the airflow required to produce each trill. This study examines this question using an airflow meter. Five speakers of Czech produced /r̝/ and /r/ in the real words řád "order," pařát "talon," tvař "face," rád "like," paráda "great," and tvar "shape." Airflow data were recorded using Macquirer. The data indicate a higher airflow during the production of /r̝/ compared to /r/. /r̝/ was produced with approximately 3 l/s more than /r/. The increased airflow is necessary to cross the boundary of laminar flow into turbulent flow and supports previous findings that /r̝/ is produced with breathy voice, which facilities trilling during frication. The data also suggests that one of the factors that makes the plain trill /r/ difficult to produce is that the airflow required to produce a sonorous trill is tightly constrained. The boundaries between trill production and the production of frication are only a few l/s apart and thus requires careful management of the laryngeal mechanisms, which control airflow.

2aSC10. Comparison of tidal breathing and reiterant speech breathing using whole body plethysmography. Marguerite Bonadies, Robert H. Withnell, and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 505 W Lava Way, Apt. C, Bloomington, IN 47404, mcbonadi@umail.iu.edu)

Classic research in the field of speech breathing has found differences in the characteristics of breathing patterns between speech respiration and tidal breathing. Though much research has been done on speech breathing mechanisms, relatively little research has been done using the whole body plethysmograph. In this study, we sought to examine differences and similarities between tidal respiration and breathing in reiterant speech using measures obtained through whole-body plethysmography. We hypothesize that there are not significant differences between pulmonary measures in tidal respiration and in speech breathing. This study involves tidal breathing on a spirometer attached to the whole-body plethysmograph followed by reiterant speech using the syllable /da/ while reading the first part of The Rainbow Passage. Experimental measures include compression volumes during both breathing tasks, and absolute lung volumes as determined from the spirometer and calibrated whole-body plethysmograph. These are compared with the pulmonary subdivisions obtained from pulmonary function tests, including vital capacity, functional residual capacity, and total lung volume.

2aSC11. An electroglottography examination of fricative and sonorous segments. Phil Howson (The Univ. of Toronto, 644B-60 Harbord St., Toronto, ON M5S3L1, Canada, phil.howson@mail.utoronto.ca)

It has been previously suggested that fricative production is marked by a longer glottal opening as compared to sonorous segments. The present study uses electroglottography (EGG) and acoustic measurements to test this hypothesis by examining the activity of the vocal cords during the articulation of fricative and sonorant segments of English and Sorbian. An English and a Sorbian speakers' extended individual productions of the phonemes /s, z, ʃ, ʒ, m, n, r, l, a/ and each phoneme in the context #Ca were recorded. The open quotient was calculated using MATLAB. H1-H2 measures were taken at 5% into the vowel following each C and at 50% into the vowel. The results indicate that the glottis is open longer during the production of fricatives than for sonorous segments. Furthermore, the glottis is slightly more open for the production of nasals and liquids than it is for vowels. These results suggest that a longer glottal opening facilitates the increased airflow required to produce frication. This contrasts previous analyses which suggested that frication is primarily achieved through a tightened constriction. While a tighter constriction may be necessary, the increased airflow velocity produced by a longer glottal opening is critical for the production of frication.

2aSC12. SIPMI: Superimposing palatal profile from maxillary impression onto midsagittal articulographic data. Wei-rong Chen and Yueh-chin Chang (Graduate Inst. of Linguist, National Tsing Hua Univ., 2F-5, No. 62, Ln. 408, Zhong-hua Rd., Zhubei City, Hsinchu County-302, Taiwan, waitlong75@gmail.com)

Palatal traces reconstructed by current advanced technologies of real-time mid-sagittal articulatory tracking (e.g., EMA, ultrasound, rtMRI, etc.) are mostly in low-resolution and lack concrete anatomical/orthodontic reference points as firm articulatory landmarks for determining places of articulation. The present study proposes a method of superimposing a physical palatal profile extracted from maxillary impression, onto mid-sagittal articulatory data. The whole palatal/dental profile is first obtained from performing an alginate maxillary impression, and a plaster maxillary mold is made from the impression. Then, the mold is either (1) cut into halves for hand-tracing or (2) 3D-scanned to extract a high resolution mid-sagittal palatal line. The mid-sagittal palatal line made from maxillary mold is further subdivided into articulatory zones, following definitions of articulatory landmarks in the literature (e.g., Catford 1988), by referring to anatomical/orthodontic landmarks imprinted on the mold. Lastly, the high-resolution, articulatorily divided palatal line can be superimposed, by using modified Iterative Closet Point (ICP) algorithm, onto the reconstructed, low-resolution palatal traces in the real-time mid-sagittal articulatory data, so that clearly divided places of articulation on palate can be visualized with articulatory movements. Evaluation results show that both hand-tracing and 3D-scanned palatal profiles yield accurate superimpositions and satisfactory visualizations of place of articulation in our EMA data.

2aSC13. Waveform morphology of pre-speech brain electrical potentials. Silas Smith and Al Yonovitz (Dept. of Commun. Sci. and Disord., The Univ. of Montana, The Univ of Montana, Missoula, MT 59812, silas.smith@umconnect.umt.edu)

The inter- and intra-subject variations of the cortical responses before the initiation of speech were recorded. These evoked potentials were obtained at a sufficient sample rate that both slow negative waves as well as faster neurogenic signals were obtained. 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. Subjects were 12 males and females. Electrodes were recorded with a silver–silver chloride electrodes positioned at Cz and using the earlobes as reference and ground. A biological preamplifier was used to amplify the weak bioelectric signals 100,000 times. Each time epoch was sampled at 20,000 samples/sec. The frequency response of these amplifiers had a high-pass of 0.1 Hz and a low-pass of 3 kHz. One second of these signals were averaged for 100 trials just prior to the subject initiation of the word "pool." Electrical brain potentials have proven to be extremely useful for diagnosis, treatment, and research in the auditory system, and are expected to be of equal importance for the speech system.

2aSC14. Acoustic correlates of bilingualism: Relating phonetic production to language experience and attitudes. Wai Ling Law (Linguist, Purdue Univ., Beering Hall, 00 North University St., West Lafayette, IN 47907, wlaw@purdue.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Researchers tend to quantify degree of bilingualism according to age-related factors such as age of acquisition (Flege, *et al*. 1999, Yeni-Komshian, *et al*. 2000). However, previous research suggests that bilinguals may also show different degrees of accent and patterns of phonetic interaction between their first language (L1) and second language (L2) as a result of factors such as the quantity and quality of L2 input (Flege & Liu, 2001), amount of L1 vs. L2 use (Flege, *et al*. 1999), and attitude toward each language (Moyer, 2007). The goal of this study is to identify gradient properties of speech production that can be related to gradient language experience and attitudes in a bilingual population that is relatively homogeneous in terms of age-related factors. Native Cantonese-English bilinguals living in Hong Kong produced near homophones in both languages under conditions emphasizing one language or the other on different days. Acoustic phonetic variables related to phonological inventory differences between the two languages, including lexical tone/stress, syllable length, nasality, fricative manner and voicing, release of stop, voice onset time, and vowel quality and length, will be quantified and compared to results from a detailed survey of individual speakers' experience and attitudes toward the two languages.

2aSC15. Dialectal variation in affricate place of articulation in Korean. Yoonjnung Kang (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, HW314, Toronto, ON M1C 1A4, Canada, yoonjung.kang@utoronto.ca), Sungwoo Han (Dept. of Korean Lang. and Lit., Inha Univ., Incheon, South Korea), Alexei Kochetov (Dept. of Linguist, Univ. of Toronto, Toronto, ON, Canada), and Eunjong Kong (Dept. of English, Korea Aerosp. Univ., Goyang, South Korea)

The place of articulation (POA) of Korean affricates has been a topic of much discussion in Korean linguistics. The traditional view is that the affricates were dental in the 15th century and then changed to a posterior coronal place in most dialects of Korean but the anterior articulation is retained in major dialects of North Korea, most notably Phyengan and Yukjin. However, recent instrumental studies on Seoul Korean and some impressionistic descriptions of North Korean dialects cast doubt on the validity of this traditional view. Our study examines the POA of /c/ (lenis affricate) and /s/ (anterior fricative) before /a/ in Seoul Korean (26 younger and 32 older speakers) and in two North Korean varieties, as spoken by ethnic Koreans in China (14 Phyengan and 21 Yukjin speakers). The centre of gravity of the frication noise of /c/ and /s/ was examined. The results show that in both North Korean varieties, both sibilants are produced as anterior coronal and comparable in their POA. In Seoul Korean, while the POA contrast shows a significant interaction with age and gender, the affricate is consistently and substantially more posterior than the anterior fricative across all speaker groups. The results support the traditional description.

2aSC16. An articulatory study of high vowels in Mandarin produced by native and non-native speakers. Chenhuei Wu (Dept. of Chinese Lang. and Lit., National Hsinchu Univ. of Education, No. 521, Nanda Rd, Hsinchu 300, Taiwan, chenhueiwu@gmail.com), Weirong Chen (Graduate Inst. of Linguist, National Tsing-hua Univ., Hsinchu, Taiwan), and Chilin Shih (Dept. of Linguist, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

This paper examined the articulatory properties of high vowels [i], [y], and [u] in Mandarin produced by four Taiwanese Mandarin native speakers and four English-speaking Chinese learners (L2 learners) with an Electromagnetic Articulagroph AG500. The articulatory positions of the tongue top (TT), the tongue body (TB), the tongue dorsal (TD), and the lips were investigated. The TT, TB, and TD of [y] produced by the L2 learner were further back than that by the native. In addition, the TD of [y] by the L2 learners was higher than the native. Further comparison found that the tongue positions of [y] was similar to [u] in L2 production. Regarding to the lip positions, the [y] and [u] were more protruded that [i] in the native production, while there is no difference among these three vowels in the L2 production.

The findings suggested that most of the L2 learner were not aware that the lingual target [y] should be very similar to [i] but the lip articulators of [y] are more protruded than [i]. Some L2 learners pronounce [y] more like a diphthong [iu] rather than a monophthong.

2aSC17. Production and perception training of /r l/ with native Japanese speakers. Anna M. Schmidt (School of Speech Path. & Aud., Kent State Univ., A104 MSP, Kent, OH 444242, aschmidt@kent.edu)

Visual feedback with electropalatometry was used to teach accurate /r/ and /l/ to a native Japanese speaker. Perceptual differentiation of the phonemes did not improve. A new perceptual training protocol was developed and tested.

2aSC18. Production of a non-phonemic variant in a second language: Acoustic analysis of Japanese speakers' production of American English flap. Mafuyu Kitahara (School of Law, Waseda Univ., 1-6-1 Nishiwaseda, Shinjuku-ku, Tokyo 1698050, Japan, kitahara@waseda.jp), Keiichi Tajima (Dept. of Psych., Hosei Univ., Tokyo, Japan), and Kiyoko Yoneyama (Dept. of English Lang., Daito Bunka Univ., Tokyo, Japan)

Second-language (L2) learners need to learn the sound system of an L2 so that they can distinguish L2 words. However, it is also instructive to learn non-phonemic, allophonic variations, particularly if learners want to sound native-like. The production of intervocalic /t d/ as an alveolar flap is a prime example of a non-phonemic variation that is salient in American English and presumably noticeable to many L2 learners. Yet, how well such non-phonemic variants are learned by L2 learners is a relatively under-explored subject. In the present study, Japanese learners' production of alveolar flaps was investigated, to clarify how well learners can learn the phonetic environments in which flapping tends to occur, and how L2 experience affects their performance. Native Japanese speakers who had lived in North America for various lengths of time read a list of words and phrases that contained a potentially flappable stop, embedded in a carrier sentence. Preliminary results indicated that the rate of flapping varied considerably across different words and phrases and across speakers. Furthermore, acoustic parameters such as flap closure duration produced by some speakers showed intermediate values between native-like flaps and regular stops, suggesting that flapping is a gradient phenomenon. [Work supported by JSPS.]

2aSC19. A comparison of speaking rate consistency in native and non-native speakers of English. Melissa M. Baese-Berk (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu) and Tuuli Morrill (Linguist, George Mason Univ., Fairfax, VA)

Non-native speech differs from native speech in many ways, including overall longer durations and slower speech rates (Guion *et al*., 2000). Speaking rate also influences how listeners perceive speech, including perceived fluency of non-native speakers (Munro & Derwing, 1998). However, it is unclear what aspects of non-native speech and speaking rate might influence perceived fluency. It is possible that in addition to differences in mean speaking rate, there may be differences in the consistency of speaking rate within and across utterances. In the current study, we use production data to examine speaking rate in native and non-native speakers of English, and ask whether native and non-native speakers differ in the consistency of their speaking rate across and within utterances. We examined a corpus of read speech, including isolated sentences and longer narrative passages. Specifically, we test whether the overall slower speech rate of non-native speakers is coupled with an inconsistent speech rate that may result in less predictability in the produced speech signal.

2aSC20. Relative distances among English front vowels produced by Korean and American speakers. Byunggon Yang (English Education, Pusan National Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

This study examined the relative distances among English front vowels in a message produced by 47 Korean and American speakers from an internet speech archive in order to provide better pronunciation skills for Korean English learners. The Euclidean distances in the vowel space of F1 and F2 were measured among the front vowel pairs. The first vowel pair [i-ɛ] was set as the reference from which the relative distances of the other two vowel pairs were measured in percent in order to compare the vowel sounds among speakers of different vocal tract lengths. Results show that F1 values of the front vowels produced by the Korean and American speakers increased from the high front vowel to the low front vowel with differences among the groups. The Korean speakers generally produced the front vowels with smaller jaw openings than the American speakers did. Second, the relative distance of the high front vowel pair [i-ɪ] showed a significant difference between the Korean and American speakers while that of the low front vowel pair [ɛ-æ] showed a non-significant difference. Finally, the Korean speakers in the higher proficiency level produced the front vowels with higher F1 values than those in the lower proficiency level.

Session 2aUW

Underwater Acoustics: Signal Processing and Ambient Noise

Jorge E. Quijano, Chair
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Contributed Papers

8:00

2aUW1. Moving source localization and tracking based on data. Tsih C. Yang (Inst. of Undersea Technol., National Sun Yat-sen Univ., 70 Lien Hai Rd., Kaohsiung 80404, Taiwan, tsihyang@gmail.com)

Matched field processing (MFP) was introduced sometimes ago for source localization based on the replica field for a hypothesized source location that best matches the acoustic data received on a vertical line array (VLA). A data-based matched-mode source localization method is introduced in this paper for a moving source, using mode wavenumbers and depth functions estimated directly from the data, without requiring any environmental acoustic information and assuming any propagation model to calculate the replica field. The method is in theory free of the environmental mismatch problem since the mode replicas are estimated from the same data used to localize the source. Besides the estimation error due to the approximations made in deriving the data-based algorithms, the method has some inherent drawbacks: (1) it uses a smaller number of modes than theoretically possible, since some modes are not resolved in the measurements, and (2) the depth search is limited to the depth covered by the receivers. Using simulated data, it is found that the performance degradation due to the above approximation/limitation is marginal compared with the original matched-mode source localization method. Certain aspects of the proposed method have previously been tested against data. The key issues are discussed in this paper.

8:15

2aUW2. Simultaneous localization of multiple vocalizing humpback whale calls in an ocean waveguide with a single horizontal array using the array invariant. Zheng Gong, Sunwoong Lee (Mech. Eng., Massachusetts Inst. of Technol., 5-435, 77 Massachusetts Ave., Cambridge, MA 02139, zgong@mit.edu), Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), and Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

The array invariant method, previously derived for instantaneous range and bearing estimation of a broadband impulsive source in a horizontally stratified ocean waveguide [Lee and Makris, J. Acoust. Soc. Am. 119, 336–351 (2006)], is generalized to instantaneously and simultaneously localize multiple uncorrelated broadband noise sources that not necessarily impulsive in the time domain. In an ideal Pekeris waveguide, we theoretically show that source range and bearing can be instantaneously obtained from beam-time migration lines measured with a horizontal array through range and bearing dependent differences that arise between modal group speeds along the array. We also show that this theory is approximately valid in a horizontally stratified ocean waveguide. A transform, similar to the Radon transform, is employed to enable simultaneous localization of multiple uncorrelated broadband noise sources without ambiguity using the array invariant method. The method is now applied to humpback whale vocalization data from the Gulf of Maine 2006 Experiment for humpback whale ranges up to tens of kilometers, where it is shown that accurate bearing and range estimation of multiple vocalizing humpback whales can be simultaneously made with little computational effort.

8:30

2aUW3. Test for eigenspace stationarity applied to multi-rate adaptive beamformer. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, jorgess39@hotmail.com) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR)

Array processing in the presence of moving targets is challenging since the number of stationary data snapshots required for estimation of the data covariance are limited. For experimental scenarios that include a combination of fast maneuvering loud interferers and quiet targets, the multi-rate adaptive beamformer (MRABF) can mitigate the effect of non-stationarity. In MRABF, the eigenspace associated to loud interferers is first estimated and removed, followed by application of adaptive beamforming techniques to the remaining, less variable, "target" subspace. Selection of the number of snapshots used for estimation of the interferer eigenspace is crucial in the operation of MRABF, since too few snapshots result in poor eigenspace estimation, while too many snapshots result in leakage of non-stationary interferer effects into the target subspace. In this work an eigenvector-based test for data stationarity, recently developed in the context of very large arrays with snapshot deficiency, is used as a quantitative method to select the optimal number of snapshots for the estimation of the non-stationary eigenspace. The approach is demonstrated with simulated and experimental data from the Shallow Water Array Performance (SWAP) experiment.

8:45

2aUW4. Design of a coprime array for the North Elba sea trial. Vaibhav Chavali, Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., 4307 Ramona Dr., Apt. # H, Fairfax, VA 22030, vchavali@gmu.edu), and John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Vaidyanathan and Pal [IEEE Trans. Signal Process. 2011] proposed the use of Coprime Sensor Arrays (CSAs) to sample spatial fields using fewer elements than a Uniform Line Array (ULA) spanning the same aperture. A CSA consists of two interleaved uniform subarrays that are undersampled by coprime factors M and N. The subarrays are processed independently and then their scanned responses are multiplied to obtain a unaliased output. Although the CSA achieves resolution comparable to that of a fully populated ULA, the CSA beampattern has higher sidelobes. Adhikari *et al.* [Proc. ICASSP, 2013] showed that extending the subarrays and applying spatial tapers could reduce CSA sidelobes. This paper considers the problem of designing a CSA for the North Elba Sea Trial described by Gingras [SACLANT Tech. Report, 1994]. The experimental dataset consists of receptions recorded by a 48-element vertical ULA in a shallow water environment for two different source frequencies: 170 Hz and 335 Hz. This paper considers all possible coprime subsamplings for this array and selects the configuration that provides the best tradeoff between number of sensors and performance. Results are shown for both simulated and experimental data. [Work supported by ONR Basic Research Challenge Program.]

9:00

2aUW5. Localization of a high frequency source in a shallow ocean sound channel using frequency-difference matched field processing. Brian Worthmann (Appl. Phys., Univ. of Michigan, 3385 Oakwood St., Ann Arbor, MI 48104, bworthma@umich.edu), H. C. Song (Marine Physical Lab., Scripps Inst. for Oceanogr., Univ. of California - San Diego, La Jolla, CA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Matched field processing (MFP) is an established technique for locating remote acoustic sources in known environments. Unfortunately, environment-to-propagation model mismatch prevents successful application of MFP in many circumstances, especially those involving high frequency signals. For beamforming applications, this problem was found to be mitigated through the use of a nonlinear array-signal-processing technique called frequency difference beamforming (Abadi *et al*. 2012). Building on that work, this nonlinear technique was extended to MFP, where Bartlett ambiguity surfaces were calculated at frequencies two orders of magnitude lower than the propagated signal, where the detrimental effects of environmental mismatch are much reduced. In the Kauai Acomms MURI 2011 (KAM11) experiment, underwater signals of frequency 11.2 kHz to 32.8 kHz were broadcast 3 km through a 106-m-deep shallow-ocean sound channel and were recorded by a sparse 16-element vertical array. Using the ray-tracing code Bellhop as the propagation model, frequency difference MFP was performed, and some degree of success was found in localizing the high frequency source. In this presentation, the frequency difference MFP technique is explained, and comparisons of this nonlinear MFP technique with conventional Bartlett MFP using both simulations and KAM11 experimental data are provided. [Sponsored by the Office of Naval Research.]

9:15

2aUW6. Transarctic acoustic telemetry. Hee-Chun Song (SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu), Peter Mikhalvesky (Leidos Holdings, Inc., Arlington, VA), and Arthur Baggeroer (Mech. Eng., MIT, Cambridge, MA)

On April 9 and 13, 1999, two Arctic Climate Observation using Underwater Sound (ACOUS) tomography signals were transmitted from a 20.5-Hz acoustic source moored at the Franz Victoria Strait to an 8-element, 525-m vertical array at ice camp APLIS in the Chukchi Sea at a distance of approximately 2720 km. The transmitted signal was a 20-min long, 255-digit m-sequence that can be treated as a binary-phase shift-keying communication signal with a data rate of 2 bits/s. The almost error-free performance using either spatial diversity (three elements) for a single transmission or temporal diversity (two transmissions) with a single element demonstrates the feasibility of ice-covered trans-Arctic acoustic communications.

9:30

2aUW7. Performance of adaptive multichannel decision-feedback equalization in the simulated underwater acoustic channel. Xueli Sheng, Lina Fan (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin Eng. University Shuisheng Bldg. 803, Nantong St. 145, Harbin, Heilongjiang 150001, China, shengxueli@aliyun.com), Aijun Song, and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE)

Adaptive multichannel decision feedback equalization [M. Stojanovic, J. Catipovic, and J. G. Proakis, J. Acoust. Soc. Am. 94, 1621–1631 (1993)] is widely adopted to address the severe inter-symbol interference encountered in the underwater acoustic communication channel. In this presentation, its performance will be reported in the simulated communication channel provided by a ray-based acoustic model, for different ocean conditions and source-receiver geometries. The ray model uses the Rayleigh parameter to prescribe the sea surface effects on the acoustic signal. It also supports different types of sediment. The ray model output has been compared with the experimental data and shows comparable results in transmission loss. We will also compare against the performance of multichannel decision feedback equalization supported by existing ray models, for example, BELLHOP.

9:45

2aUW8. Space-time block code with equalization technology for underwater acoustic channels. Chunhui Wang, Xueli Sheng, Lina Fan, Jia Lu, and Weijia Dong (Sci. and Technol. on Underwater Acoust. Lab., College of Underwater Acoust. Engineering, Harbin Eng. Univ., Harbin 150001, China, 740443619@qq.com)

In order to combat the effects of multipath interference and fading for underwater acoustic (UWA) channels, this paper investigates a scheme of the combination of space-time block code (STBC) and equalization technology. STBC is used in this scheme to reduce the effects of fading for the UWA channels, then equalization technology is used in this scheme to mitigate intersymbol interference. The performance of the scheme is analyzed with Alamouti space-time coding for the UWA channels. Our simulations indicate that the combination of STBC and equalization technology provides lower bit error rates.

10:00–10:15 Break

10:15

2aUW9. Robust focusing in time-reversal mirror with a virtual source array. Gi Hoon Byun and Jea Soo Kim (Ocean Eng., Korea Maritime and Ocean Univ., Dongsam 2-dong, Yeongdo-gu, Busan, Korea, Busan, South Korea, knitpia77@gmail.com)

The effectiveness of Time-Reversal (TR) focusing has been demonstrated in various fields of ocean acoustics. In TR focusing, a probe source is required for a coherent acoustic focus at the original probe source location. Recently, the need of a probe source has been partially relaxed by introduction of the concept of a Virtual Source Array (VSA) [S. C. Walker, Philippe Roux, and W. A. Kuperman, J. Acoust. Soc. Am. 125(6), 3828–3834 (2009)]. In this study, Adaptive Time-Reversal Mirror (ATRM) based on multiple constraint method [J. S. Kim, H. C. Song, and W. A. Kuperman, J. Acoust. Soc. Am. 109(5), 1817–1825 (2001)] and Singular Value Decomposition (SVD) method are applied to a VSA for robust focusing. The numerical simulation results are presented and discussed.

10:30

2aUW10. Wind generated ocean noise in deep sea. Fenghua Li and Jingyan Wang (State Key Lab. of Acoust., Inst. of Acoust., CAS, No. 21 Beisihuanxi Rd., Beijing 100190, China, lfh@mail.ioa.ac.cn)

Ocean noise is an important topic in underwater acoustics, which has been paid much attention in last decades. Ocean noise sources may consist of wind, biological sources, ships, earthquakes and so on. This paper discusses measurements of the ocean noise intensity in deep sea during strong wind periods. During the experiment, shipping density is small enough and the wind generated noise is believed to be the dominated effect in the observed frequency range. The analyses of the recoded noise data reveal that the wind generated noise source has a strong dependence on the wind speed and frequency. Based on the data, a wind generated noise source model is presented. [Work supported by National Natural Science Foundation of China, Grant No. 11125420.]

10:45

2aUW11. Ocean ambient noise in the North Atlantic during 1966 and 2013–2014. Ana Sirovic, Sean M. Wiggins, John A. Hildebrand (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr. MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), and Mark A. McDonald (Whale Acoust., Bellvue, CO)

Low-frequency ocean ambient noise has been increasing in many parts of the worlds' oceans as a result of increased shipping. Calibrated passive acoustic recordings were collected from June 2013 to March 2014 on the south side of Bermuda in the North Atlantic, at a location where ambient noise data were collected in 1966. Monthly and hourly mean power spectra (15–1000 Hz) were calculated, in addition to skewness, kurtosis, and percentile distributions. Average spectrum levels at 40 Hz, representing shipping noise, ranged from 78 to 80 dB re: 1 $\mu Pa^2/Hz$, with a peak in March and minimum in July and August. Values recorded during this recent period were similar to those recorded during 1966. This is different from trends

observed in the Northern Pacific, where ocean ambient noise has been increasing; however, the location of this monitoring site was exposed only to shipping lanes to the south of Bermuda. At frequencies dominated by wind and waves (500 Hz), noise levels ranged from 55 to 66 dB re: 1 $\mu Pa^2/Hz$, indicating low sea states (2–3) prevailed during the summer, and higher sea states (4–5) during the winter. Seasonally important contribution to ambient sound also came from marine mammals, such as blue and fin whales.

11:00

2aUW12. Adaptive passive fathometer processing of surface-generated noise received by Nested array. Junghun Kim and Jee W. Choi (Marine Sci. and Convergent Technol., Hanyang Univ., 1271 Sa-3-dong, Ansan 426-791, South Korea, Kimjh0927@hanyang.ac.kr)

Recently, a passive fathometer technique using surface-generated ambient noise has been applied to the estimate of bottom profile. This technique performs the beamforming of ambient noise received by a vertical line array to estimate the sub-bottom layer structure as well as water depth. In the previous works, the surface noise signal processing was performed with equally spaced line arrays and the main topic of the research was the comparison of the results estimated using several beamforming techniques. In this talk, the results estimated from the ambient noise received by the Nested vertical line array (called POEMS) which consists of the total 24-elments and four sub-bands are presented. The measurements were made on the eastern coast (East Sea) of Korea. Four kinds of beamforming algorithms are applied to each sub-band and also, nested array processing combining each sub-band signal was performed to obtain the best result. The results are compared to the bottom profiles from the chirp sonar. [This research was supported by the Agency for Defense Development, Korea.]

11:15

2aUW13. Feasibility of low-frequency acoustic thermometry using deep ocean ambient noise in the Atlantic, Pacific, and Indian Oceans. Katherine F. Woolfe and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 672 Brookline St SW, Atlanta, GA 30310, katherine.woolfe@gmail.com)

Previous work has demonstrated the feasibility of passive acoustic thermometry using coherent processing of low frequency ambient noise (1–40 Hz) recorded on triangular hydrophones arrays spaced ~130 km and located in the deep sound channel. These triangular arrays are part of hydroacoustic stations of the International Monitoring System operated by the Comprehensive Nuclear Test Ban Treaty Organization (Woolfe *et al.*, J. Acoust. Soc. Am. 134, 3983). To understand how passive thermometry could potentially be extended to ocean basin scales, we present a comprehensive study of the coherent components of low-frequency ambient noise recorded on five hydroacoustic stations located Atlantic, Pacific, and Indian Oceans. The frequency dependence and seasonal variability of the spatial coherence and directionality of the low-frequency ambient noise were systematically examined at each of the tested site locations. Overall, a dominant coherent component of the low-frequency noise was found to be caused by seasonal ice-breaking events at the poles for test sites that have line-of-sight paths to polar ice. These findings could be used to guide the placement of hydrophone arrays over the globe for future long-range passive acoustic thermometry experiments.

11:30

2aUW14. Ambient noise in the Arctic Ocean measured with a drifting vertical line array. Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu), John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), and John N. Kemp (Woods Hole Oceanographic Inst., Woods Hole, MA)

In mid-April 2013, a Distributed Vertical Line Array (DVLA) with 22 hydrophone modules over a 600-m aperture immediately below the subsurface float was moored near the North Pole. The top ten hydrophones were spaced 14.5 m apart. The distances between the remaining hydrophones increased geometrically with depth. Temperature and salinity were measured by thermistors in the hydrophone modules and ten Sea-Bird MicroCATs. The mooring parted just above the anchor shortly after deployment and subsequently drifted slowly south toward Fram Strait until it was recovered in mid-September 2013. The DVLA recorded low-frequency ambient noise (1953.125 samples per second) for 108 minutes six days per week. Previously reported noise levels in the Arctic are highly variable, with periods of low noise when the wind is low and the ice is stable and periods of high noise associated with pressure ridging. The Arctic is currently undergoing dramatic changes, including reductions in the extent and thickness of the ice cover, the amount of multiyear ice, and the size of the ice keels. The ambient noise data collected as the DVLA drifted will test the hypothesis that these changes result in longer and more frequent periods of low noise conditions than experienced in the past.

Session 2pAA

Architectural Acoustics and Engineering Acoustics: Architectural Acoustics and Audio II

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

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Invited Papers

1:00

2pAA1. Defining home recording spaces. Sebastian Otero (Acustic-O, Laurel 14, San Pedro Martir, Tlalpan, Mexico, D.F. 14650, Mexico, sebastian@acustic-o.com)

The idea of home recording has been widely used through out the audio and acoustics community for some time. The effort and investment put into these projects fluctuate in such a wide spectrum that there is no clear way to unify the concept of "home studio," making it difficult for acoustical consultants and clients to reach an understanding on each other project goals. This paper analyses different spaces which vary in terms of privacy, comfort, size, audio quality, budget, type of materials, acoustic treatments, types of projects developed and equipment, but which can all be called "home recording spaces," in order to develop a more specific classification of these environments.

1:20

2pAA2. Vibrato parameterization. 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)

In an effort to improve the quality of synthetic vibrato many musical instrument tones with vibrato have been analyzed and frequency-vs-time curves have been parameterized in terms of a time-varying offset and a time-varying vibrato depth. Results for variable mean F0 and instrument are presented. Whereas vocal vibrato appears to sweep out the resonance characteristic of the vocal tract, as shown by amplitude-vs-frequency curves for the superposition of a range of harmonics, amplitude-vs-frequency curves for instruments are dominated by hysteresis effects that obscure their interpretation in terms of resonance characteristics. Nevertheless, there is a strong correlation between harmonic amplitude and frequency modulations. An effort is being made to parameterize this effect in order to provide efficient and expressive synthesis of vibrato tones with independent control of vibrato rate and tone duration.

1:40

2pAA3. Get real: Improving acoustic environments in video games. Yuri Lysoivanov (Recording Arts, Tribeca Flashpoint Media Arts Acad., 28 N. Clark St. Ste. 500, Chicago, IL 60602, yuri.lysoivanov@tfa.edu)

As processing power grows the push for realism in video games continues to expand. However, techniques for generating realistic acoustic environments in games have often been limited. Using examples from major releases, this presentation will take a historical perspective on interactive environment design, discuss current methods for modeling acoustic environments in games and suggest specific cases where acoustic expertise can provide an added layer to the interactive experience.

2:00

2pAA4. Applications of telematic mixing consoles in networked audio for musical performance, spatial audio research, and sound installations. David J. Samson and Jonas Braasch (Rensselaer Polytechnic Inst., 1521 6th Ave., Apt. 303, Troy, NY 12180, samsod2@rpi.edu)

In today's technologically driven world, the ability to connect across great distance via Internet Protocol is more important than ever. As the technology evolves, so does the art and science that relies upon it for collaboration and growth. Developing the state of the art system for flexible and efficient routing of networked audio provides a platform for experimental musicians, researchers, and artists to create freely without the restrictions imposed by traditional telepresence. Building on previous development and testing of a telematic mixing console while addressing critical issues with the current platform and current practice, the console allows for the integration of high-quality networked audio into computer assisted virtual environments (CAVE systems), sound and art installations, and other audio driven research projects. Through user study, beta testing, and integration into virtual audio environments, the console has evolved to meet the demand for power and flexibility critical to multi-site collaboration with high-quality networked audio. Areas of concern addressed in development are computational efficiency, system latency, routing architecture, and results of continued user study.

2:20

2pAA5. Twenty years of electronic architecture in the Hilbert Circle Theatre. Paul Scarbrough (Akustiks, 93 North Main St., South Norwalk, CT 06854, pscarbrough@akustiks.com) and Steve Barbar (E-coustic Systems, Belmont, MA)

In 1984, the Circle Theatre underwent a major renovation, transforming the original 3000+ seat venue into a 1780 seat hall with reclaimed internal volume dedicated to a new lobby and an orchestra rehearsal space. In 1996, a LARES acoustic enhancement system replaced the original electronic architecture system, and has been used in every performance since that time. We will discuss details of the renovation, the incorporation of the electronic architecture with other acoustical treatments, system performance over time, and plans for the future.

2:40

2pAA6. Equalization and compression—Friends or foes? Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

These two essential signal processors have overlapping capabilities. Tuning a sound system for any function requires complementary interaction between equalization and compression. The timbral impact of compression is indirect, and can be counterintuitive. A deeper understanding of compression parameters, particularly attack and release, clarifies the connection between compression and tone and makes coordination with equalization more productive.

3:00–3:15 Break

Contributed Papers

3:15

2pAA7. Analysis of room acoustical characteristics by plane wave decomposition using spherical microphone arrays. Jin Yong Jeon, Muhammad Imran, and Hansol Lim (Dept. of Architectural Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul, 133791, South Korea, jyjeon@hanyang.ac.kr)

The room acoustical characteristics have been investigated in temporal and spatial structures of room impulse responses (IRs) at different audience positions in real halls. The spherical microphone array of 32-channel is used for measurements process. Specular and diffusive reflections in IRs have been visualized in temporal domain with sound-field decomposition analysis. For plane wave decomposition, the spherical harmonics are used. The beamforming technique is also employed to make directional measurements and for the spatio-temporal characterization of sound field. The directional measurements by beamforming are performed for producing impulse responses for the different directions to characterize the sound. From the estimation of spatial characterization, the reflective surfaces of the hall are indicated as responsible for specular and diffusive reflections.

3:30

2pAA8. Comparing the acoustical nature of a compressed earth block residence to a traditional wood-framed residence. Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

Various lost, misunderstood, or abandoned materials and methods throughout history can serve as viable options in today's impasse of nature and mankind. Similar to the 19th century resurgence of concrete, there is a developing interest in earth as an architectural material capable of dealing with unexpected fluctuations and rising climate changes. Studying the acoustical nature of earthen construction can also serve as a method of application beyond aesthetics and thermal comfort. Innovations using Compressed Earth Block (CEB) have been developed and researched over the past few decades and recently the collaborative focus for a collaborative team of faculty and students at a NAAB accredited College of Architecture, an ABET accredited College of Engineering, and a local chapter of Habitat for Humanity. The multidisciplinary research project resulted in the design and simultaneous construction of both a CEB residence and a conventionally wood-framed version of equal layout, area, volume, apertures, and roof structure on adjacent sites to prove the structural, thermal, economical, and acoustical value of CEB as a viable residential building material. This paper defines acoustical measurements of both residences such as STC, OITC, TL, NC, FFT, frequency responses, and background noise levels prior to occupancy.

3:45

2pAA9. A case study of a high end residential condominium building acoustical design and field performance testing. Erik J. Ryerson and Tom Rafferty (Acoust., Shen Milsom & Wilke, LLC, 2 North Riverside Plaza, Ste. 1460, Chicago, IL 60606, eryerson@smwllc.com)

A high end multi-owner condominium building complex consists of 314 units configured in a central tower with a 39-story central core, as well as 21 and 30 story side towers. A series of project specific acoustical design considerations related to condominium unit horizontal and vertical acoustical separation as well as background noise control for building HVAC systems were developed for the project construction documents and later field tested to confirm conformance with the acoustical design criteria. This paper presents the results of these building wide field tests as well as a discussion of pitfalls encountered during design, construction, and post-construction.

4:00

2pAA10. Innovative ways to make cross laminated timber panels sound-absorptive. Banda Logawa and Murray Hodgson (Mech. Eng., Univ. of Br. Columbia, 2160-2260 West Mall, Vancouver, BC, Canada, logawa_b@yahoo.com)

Cross Laminated Timber (CLT) panels typically consist of several glued layers of wooden boards with orthogonally alternating directions. This cross-laminating process allows CLT panels to be used as load-bearing plate elements similar to concrete slabs. However, they are very sound-reflective, which can lead to concerns about acoustics. Growing interest in applications of CLT panels as building materials in North America has initiated much current research on their acoustical properties. This project is aimed at investigating ways to improve the sound-absorption characteristics of the panels by integrating arrays of Helmholtz-resonator (HR) absorbers into the panels and establishing design guidelines for CLT-HR absorber panels for various room-acoustical applications. To design the new prototype panels, several efforts have been made to measure and analyze the sound-absorption characteristics of the exposed CLT surfaces in multiple buildings in British Columbia, investigate suitable methods and locations to measure both normal and random incidence sound absorption characteristics, study the current manufacturing method of CLT panels, create acoustic models of CLT-HR absorber panels with various shapes and dimensions, and evaluate the sound absorption performance of prototype panels. This paper will report progress on this work.

4:15

2pAA11. Investigate the persistence of sound frequencies Indoor television decors. Mohsen Karami (Dept. of Media Eng., IRIB Univ., No. 8, Dahmetry 4th Alley, Bahar Ave., Kermanshah, Kermanshah 6718839497, Iran, mohsenkarami.ir@gmail.com)

It seems to add to the décor of the studio and make a half-closed spaces and reduce the absorption of waves hitting the studio sound and lasting changes in the frequency of sound energy absorption occurs. To address this issue, the pink noise playback with B&K2260 device, standard time ISO3382, reverberation time in IRIB channels programs twelve decors was measured in various studios. Survey shows values obtained in all the décor, the persistence of high frequencies and this effect occurred regardless of the décor's shape and the studio is.

TUESDAY AFTERNOON, 28 OCTOBER 2014 LINCOLN, 1:25 P.M. TO 5:00 P.M.

Session 2pAB

Animal Bioacoustics: Topics in Animal Bioacoustics II

Cynthia F. Moss, Chair
Psychological and Brain Sci., Johns Hopkins Univ., 3400 N. Charles St., Ames Hall 200B, Baltimore, MD 21218

Chair's Introduction—1:25

Contributed Papers

1:30

2pAB1. Amplitude shifts in the cochlear microphonics of Mongolian gerbils created by noise exposure. Shin Kawai and Hiroshi Riquimaroux (Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

The Mongolian gerbil (Meriones unguiculatus) was used to evaluate effects of intense noise exposure on functions of the hair cells. Cochlear microphonics (CM) served an index to show functions of the hair cells. The purpose of this study was to verify which frequency was most damaged by noise exposure and examine relationships between the frequency and the animal's behaviors. We measured growth and recovery of the temporal shifts in amplitude of CM. The CM was recorded from the round window. Test stimuli used were tone bursts (1–45 kHz in half octave step), with duration of 50 ms (5 ms rise/fall times). The subject was exposed to broadband noise (0.5 to 60 kHz) at 90 dB SPL for 5 minutes. Threshold shifts were measured for the testing tone bursts from immediately after the exposure up to 120 minutes after the exposure. Findings showed that reduction in CM amplitude was observed after the noise exposure. Especially, large reduction was produced in a frequency range of 22.4 kHz. However, little reduction was observed around a frequency range of 4 kHz.

1:45

2pAB2. Detection of fish calls by using the small underwater sound recorder. Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@cs.tohoku-gakuin.ac.jp), Tomohito Imaizumi, and Tmonari Akamatsu (National Res. Inst. of Fisheries Eng., Fisheries Res. Agency, Kamisu, Japan)

Passive acoustic monitoring has been widely used for the survey of marine mammals. This method can be applied for any sounding creatures in the ocean. Many fish, including croaker, grunt, and snapper, produce species-specific low-frequency sounds associated with courtship and spawning behavior in chorus groups. In this paper, the acoustic data accumulated by an autonomous small underwater recorder were used for the sound detection analysis. The recorder was set on the sea floor off the coast of Chosi in Japan (35°40′55″N, 140°49′14″E). The observed signals include not only target fish calls (white croaker) but also calls of another marine life and noises of vessels. We tried to extract the target fish calls out of the sounds. First, recordings were processed by bandpass filter (400–2400 Hz) to eliminate low frequency noise contamination. Second, a low frequency filter applied to extract envelope of the waveform and identified high intensity sound units, which are possibly fish calls. Third, parameter tuning has been conducted to fit the detection of target fish call using absolute received intensity and duration. In this method, 28614 fish calls could be detected from the observed signals during 130 hours Comparing with manually identified fish call, correct detection and false alarm were 0.88 and 0.03, respectively. [This work was supported by CREST, JST.]

2:00

2pAB3. Changes in note order stereotypy during learning in two species of songbird, measured with automatic note classification. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

In addition to mastering the task of performing the individual notes of a song, many songbirds must also learn to produce each note in a stereotyped order. As a bird practices its new song, it may perform thousands of bouts, providing a rich source of information about how note phonology and note type order change during learning. A combination of acoustic landmark descriptions, neural network and hierarchical clustering classifiers, and Markov models of note order made it possible to measure note order stereotypy in two species of songbird. Captive swamp sparrows (*Melospiza melodia*, 11 birds, 92063 notes/bird), and wild tree swallows (*Tachycineta bicolor*, 18 birds, 448 syllables/bird) were recorded song development. The predictability of swamp sparrow note order showed significant increase during the month-long recording period ($F1,162 = 9977$, $p < 0.001$). Note order stereotypy in tree swallows also increased by a significant amount over a month-long field season (Mann-Whitney $V = 12$, p-value < 0.001). Understanding changes in song stereotypy can improve our knowledge of vocal learning, performance, and cultural transmission.

2:15

2pAB4. Plugin architecture for creating algorithms for bioacoustic signal processing software. Christopher A. Marsh, Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720, cmarsh@rohan.sdsu.edu), and David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR)

There are several acoustic monitoring software packages that allow for the creation and execution of algorithms that automate detection, classification, and localization (DCL). Algorithms written for one program are generally not portable to other programs, and usually must be written in a specific programming language. We have developed an application programming interface (API) that seeks to resolve these issues by providing a plugin framework for creating algorithms for two acoustic monitoring packages: Ishmael and PAMGuard. This API will allow new detection, classification, and localization algorithms to be written for these programs without requiring knowledge of the monitoring software's source code or inner workings, and lets a single implementation run on either platform. The API also allows users to write DCL algorithms in a wide variety of languages. We hope that this will promote the sharing and reuse of algorithm code. [Funding from ONR.]

2:30

2pAB5. Acoustic detection of migrating gray, humpback, and blue whales in the coastal, northeast Pacific. Regina A. Guazzo, John A. Hildebrand, and Sean M. Wiggins (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9450 Gilman Dr., #80237, La Jolla, CA 92092, rguazzo@ucsd.edu)

Many large cetaceans of suborder Mysticeti make long annual migrations along the California coast. A bottom-mounted hydrophone was deployed in shallow water off the coast of central California and recorded during November 2012 to September 2013. The recording was used to determine the presence of blue whales, humpback whales, and gray whales. Gray whale calls were further analyzed and the number of calls per day and per hour were calculated. It was found that gray whales make their migratory M3 calls at a higher rate than previously observed. There were also more M3 calls recorded at night than during the day. This work will be continued to study the patterns and interactions between species and compared with shore-based survey data.

2:45

2pAB6. Importing acoustic metadata into the Tethys scientific workbench/database. Sean T. Herbert (Marine Physical Lab., Scripps Inst. of Oceanogr., 8237 Lapiz Dr., San Diego, CA 92126, sth.email@gmail.com) and Marie A. Roch (Comput. Sci., San Diego State Univ., San Diego, CA)

Tethys is a temporal-spatial scientific workbench/database created to enable the aggregation and analysis of acoustic metadata from recordings such as animal detections and localizations. Tethys stores data in a specific format and structure, but researchers produce and store data in various formats. Examples of storage formats include spreadsheets, relational databases, or comma-separated value (CSV) text files. Thus, one aspect of the Tethys project has been to focus on providing options to allow data import regardless of the format in which it is stored. Data import can be accomplished in one of two ways. The first is translation, which transforms source data from other formats into the format Tethys uses. Translation does not require any programming, but rather the specification of an import map which associates the researcher's data with Tethys fields. The second method is a framework called Nilus that enables detection and localization algorithms to create Tethys formatted documents directly. Programs can either be designed around Nilus, or be modified to make use of it, which does require some programming. These two methods have been used to successfully import over 4.5 million records into Tethys. [Work funded by NOPP/ONR/BOEM.]

3:00–3:15 Break

3:15

2pAB7. Temporal patterns in detections of sperm whales (Physeter macrocephalus) in the North Pacific Ocean based on long-term passive acoustic monitoring. Karlina Merkens (Protected Species Div., NOAA Pacific Islands Fisheries Sci. Ctr., NMFS/PIFSC/PSD/Karlina Merkens, 1845 Wasp Blvd., Bldg. 176, Honolulu, HI 96818, karlina.merkens@noaa.gov), Anne Simonis (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Erin Oleson (Protected Species Div., NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI)

Sperm whales (Physeter macrocephalus), a long-lived, cosmopolitan species, are well suited for long-term studies, and their high amplitude echolocation signals make them ideal for passive acoustic monitoring. NOAA's Pacific Islands Fisheries Science Center has deployed High-frequency Acoustic Recording Packages (200 kHz sampling rate) at 13 deep-water locations across the central and western North Pacific Ocean since 2005. Recordings from all sites were manually analyzed for sperm whale signals, and temporal patterns were examined on multiple scales. There were sperm whale detections at all sites, although the rate of detection varied by location, with the highest rate at Wake Island (15% of samples), and the fewest detections at sites close to the equator (<1%). Only two locations (Saipan and Pearl and Hermes Reef) showed significant seasonal patterns, with more detections in the early spring and summer than in later summer or fall. There were no significant patterns relating to lunar cycles. Analysis of diel variation revealed that sperm whales were detected more during the day and night compared to dawn and dusk at most sites. The variability shown in these results emphasizes the importance of assessing basic biological patterns and variations in the probability of detection before progressing to further analysis, such as density estimation, where the effects of uneven sampling effort could significantly influence results.

3:30

2pAB8. Automatic detection of tropical fish calls recorded on moored acoustic recording platforms. Maxwell B. Kaplan, T. A. Mooney (Biology, Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS50, Woods Hole, MA 02543, mkaplan@whoi.edu), and Jim Partan (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Passive acoustic recording of biological sound production on coral reefs can help identify spatial and temporal differences among reefs; however, the contributions of individual fish calls to overall trends are often overlooked. Given that the diversity of fish call types may be indicative of fish species diversity on a reef, quantifying these call types could be used as a proxy measure for biodiversity. Accordingly, automatic fish call detectors are needed because long acoustic recorders deployments can generate large volumes of data. In this investigation, we report the development and performance of two detectors—an entropy detector, which identifies troughs in entropy (i.e., uneven distribution of entropy across the frequency band of interest, 100–1000 Hz), and an energy detector, which identifies peaks in root mean square sound pressure level. Performance of these algorithms is assessed against a human identification of fish sounds recorded on a coral reef in the US Virgin Islands in 2013. Results indicate that the entropy and energy detectors, respectively, have false positive rates of 9.9% and 9.9% with false negative rates of 28.8% and 31.3%. These detections can be used to cluster calls into types, in order to assess call type diversity at different reefs.

3:45

2pAB9. Social calling behavior in Southeast Alaskan humpback whales (Megaptera novaeangliae): Communication and context. Michelle Fournet (Dept. of Fisheries and Wildlife, Oregon State Univ., 425 SE Bridgeway Ave., Corvallis, OR 97333, mbellalady@gmail.com), Andrew R. Szabo (Alaska Whale Foundation, Petersburg, AK), and David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR)

Across their range humpback whales (Megaptera novaeangliae) produce a wide array of vocalizations including song, foraging vocalizations, and a variety of non-song vocalizations known as social calls. This study investigates the social calling behavior of Southeast Alaskan humpback whales from a sample of 299 vocalizations paired with 365 visual surveys collected

over a three-month period on a foraging ground in Frederick Sound in Southeast Alaska. Vocalizations were classified using visual-aural analysis, statistical cluster analyses, and discriminant function analysis. The relationship between vocal behavior and spatial-behavioral context was analyzed using a Poisson log-linear regression (PLL). Preliminary results indicate that some call types were commonly produced while others were rare, and that the greatest variety of calling occurred when whales were clustered. Moreover, calling rates in one vocal class, the pulsed (P) vocal class, were negatively correlated with mean nearest-neighbor distance, indicating that P calling rates increased as animals clustered, suggesting that the use of P calls may be spatially mediated. The data further suggest that vocal behavior varies based on social context, and that vocal behavior trends toward complexity as the potential for social interactions increases. [Work funded by Alaska Whale Foundation and ONR.]

4:00

2pAB10. First measurements of humpback whale song received sound levels recorded from a tagged calf. Jessica Chen, Whitlow W. L. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744, jchen2@hawaii.edu), and Adam A. Pack (Departments of Psych. and Biology, Univ. of Hawaii at Hilo, Hilo, HI)

There is increasing concern over the potential ecological effects from high levels of oceanographic anthropogenic noise on marine mammals. Current US NOAA regulations on received noise levels as well as the Draft Guidance for Assessing the Effect of Anthropogenic Sound on Marine Mammals are based on limited studies conducted on few species. For the regulations to be effective, it is important to first understand what whales hear and their received levels of natural sounds. This novel study presents the measurement of sound pressure levels of humpback whale song received at a humpback whale calf in the wintering area of Maui, Hawaii. This individual was tagged with an Acousonde acoustic and data recording tag and captured vocalizations from a singing male escort associated with the calf and its mother. Although differences in behavioral reaction to anthropogenic versus natural sounds have yet to be quantified, this represents the first known measurement of sound levels that a calf may be exposed to naturally from conspecifics. These levels can also be compared to calculated humpback song source levels. Despite its recovering population, the humpback whale is an endangered species and understanding their acoustic environment is important for continued regulation and protection.

4:15

2pAB11. Seismic airgun surveys and vessel traffic in the Fram Strait and their contribution to the polar soundscape. Sharon L. Nieukirk, Holger Klinck, David K. Mellinger, Karolin Klinck, and Robert P. Dziak (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, sharon.nieukirk@oregonstate.edu)

Low-frequency (<1 kHz) noise associated with human offshore activities has increased dramatically over the last 50 years. Of special interest are areas such as the Arctic where anthropogenic noise levels are relatively low but could change dramatically, as sea ice continues to shrink and trans-polar shipping routes open. In 2009, we began an annual deployment of two calibrated autonomous hydrophones in the Fram Strait to record underwater ambient sound continuously for one year at a sampling rate of 2 kHz. Ambient noise levels were summarized via long-term spectral average plots and reviewed for anthropogenic sources. Vessel traffic data were acquired from the Automatic Identification System (AIS) archive and ship density was estimated by weighting vessel tracklines by vessel length. Background noise levels were dominated by sounds from seismic airguns during spring, summer and fall months; during summer these sounds were recorded in all hours of the day and all days of a month. Ship density in the Fram Strait peaked in late summer and increased every year. Future increases in ship traffic and seismic surveys coincident with melting sea ice will increase ambient noise levels, potentially affecting the numerous species of acoustically active whales using this region.

4:30

2pAB12. Using the dynamics of mouth opening in echolocating bats to predict pulse parameters among individual *Eptesicus fuscus*. Laura N. Kloepper, James A. Simmons (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Brown University Providence, RI 02912, laura_kloepper@brown.edu), and John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

The big brown bat (Eptesicus fuscus) produces echolocation sounds in its larynx and emits them through its open mouth. Individual mouth-opening cycles last for about 50 ms, with the sound produced in the middle, when the mouth is approaching or reaching maximum gape angle. In previous work, the mouth gape-angle at pulse emission only weakly predicted pulse duration and the terminal frequency of the first-harmonic FM downsweep. In the present study, we investigated whether the dynamics of mouth opening around the time of pulse emission predict additional pulse waveform characteristics. Mouth angle openings for 24 ms before and 24 ms after pulse emission were compared to pulse waveform parameters for three big brown bats performing a target detection task. In general, coupling to the air through the mouth seems less important than laryngeal factors for determining acoustic parameters of the broadcasts. Differences in mouth opening dynamics and pulse parameters among individual bats highlight this relation. [Supported by NSF and ONR.]

4:45

2pAB13. Investigating whistle characteristics of three overlapping populations of false killer whales (*Pseudorca crassidens*) in the Hawaiian Islands. Yvonne M. Barkley, Erin Oleson (NOAA Pacific Islands Fisheries Sci. Ctr., 1845 Wasp Blvd., Bldg. 176, Honolulu, HI 96818, yvonne.barkley@noaa.gov), and Julie N. Oswald (Bio-Waves, Inc., Encinitas, CA)

Three genetically distinct populations of false killer whales Pseudorca crassidens) reside in the Hawaiian Archipelago: two insular populations (one within the main Hawaiian Islands [MHI] and the other within the Northwestern Hawaiian Islands [NWHI]), and a wide-ranging pelagic population with a distribution overlapping the two insular populations. The mechanisms that created and maintain the separation among these populations are unknown. To investigate the distinctiveness of whistles produced by each population, we adapted the Real-time Odontocete Call Classification Algorithm (ROCCA) whistle classifier to classify false killer whale whistles to population based on 54 whistle measurements. 911 total whistles from the three populations were included in the analysis. Results show that the MHI population is vocally distinct, with up to 80% of individual whistles correctly classified. The NWHI and pelagic populations achieved between 48 and 52% correct classification for individual whistles. We evaluated the sensitivity of the classifier to the input whistle measurements to determine which variables are driving the classification results. Understanding how these three populations differ acoustically may improve the efficacy of the classifier and create new acoustic monitoring approaches for a difficult-to-study species.

Session 2pAO

Acoustical Oceanography: General Topics in Acoustical Oceanography

John A. Colosi, Chair
Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943

Contributed Papers

1:45

2pAO1. Analysis of sound speed fluctuations in the Fram Strait near Greenland during summer 2013. Kaustubha Raghukumar, John A. Colosi (Oceanogr., Naval Postgrad. School, 315B Spanagel Hall, Monterey, CA 93943, kraghuku@nps.edu), and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

We analyze sound speed fluctuations in roughly 600 m deep polar waters from a recent experiment. The Thin-ice Arctic Acoustics Window (THAAW) experiment was conducted in the waters of Fram Strait, east of Greenland, during the summer of 2013. A drifting acoustic mooring that incorporated environmental sensors measured temperature and salinity over a period of four months, along a 500 km north-south transect. We examine the relative contributions of salinity-driven polar internal wave activity, and temperature/salinity variability along isopycnal surfaces (spice) on sound speed perturbations in the Arctic. Both internal-wave and spice effects are compared against the more general deep water PhilSea09 measurements. Additionally, internal wave spectra, energies, and modal bandwidth are compared against the well-known Garrett-Munk spectrum. Given the resurgence of interest in polar acoustics, we believe that this analysis will help parameterize sound speed fluctuations in future acoustic propagation models.

2:00

2pAO2. Sound intensity fluctuations due to mode coupling in the presence of nonlinear internal waves in shallow water. Boris Katsnelson (Marine geoSci., Univ. of Haifa, Mt Carmel, Haifa 31905, Israel, katz@phys.vsu.ru), Valery Grogirev (Phys., Voronezh Univ., Voronezh, Russian Federation), and James Lynch (WHOI, Woods Hole, MA)

Intensity fluctuations of the low frequency LFM signals (band 270–330 Hz) were observed in Shallow Water 2006 experiment in the presence of moving train consisting of about seven separate nonlinear internal waves crossing acoustic track at some angle (~ 80°). It is shown that spectrum of the sound intensity fluctuations calculated for time period of radiation (about 7.5 minutes) contains a few peaks, corresponding to predominating frequency ~6.5 cph (and its harmonics) and small peak, corresponding to comparatively high frequency, about 30 cph, which is interpreted by authors as manifestation of horizontal refraction. Values of mentioned frequencies are in accordance with theory of mode coupling and horizontal refraction on moving nonlinear internal waves, developed earlier by authors. [Work was supported by BSF.]

2:15

2pAO3. A comparison of measured and forecast acoustic propagation in a virtual denied area characterized by a heterogeneous data collection asset-network. Yong-Min Jiang and Alberto Alvarez (Res. Dept., NATO-STO-Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, La Spezia 19126, Italy, jiang@cmre.nato.int)

The fidelity of sonar performance predictions depends on the model used and the quantity and quality of the environmental information that is available. To investigate the impact of the oceanographic information collected by a heterogeneous and near-real time adaptive network of robots in a simulated access denied area, a field experiment (REP13-MED) was conducted by CMRE during August 2013 in an area (70 × 81 km) located offshore La Spezia (Italy), in the Ligurian Sea. The sonar performance assessment makes use of acoustic data recorded by a vertical line array at source—receiver ranges from 0.5 to 30 km. Continuous wave pulses at multiple frequencies (300–600 Hz) were transmitted at two source depths, 25 and 60 meters, at each range. At least 60 pings were collected for each source depth to build up the statistics of the acoustic received level and quantify the measurement uncertainty. A comparison of the acoustic transmission loss measured and predicted using an ocean prediction model (ROMS) assimilating the observed oceanographic data is presented, and the performance of the observational network is evaluated. [Work funded by NATO–Allied Command Transformation]

2:30

2pAO4. Performance assessment of a short hydrophone array for seabed characterization using natural-made ambient noise. Peter L. Nielsen (Res. Dept., STO-CMRE, V.S. Bartolomeo 400, La Spezia 19126, Italy, nielsen@cmre.nato.int), Martin Siderius, and Lanfranco Muzi (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR)

The passive acoustic estimate of seabed properties using natural-made ambient noise received on a glider equipped hydrophone array provides the capability to perform long duration seabed characterization surveys on demand in denied areas. However, short and compact arrays associated with gliders are limited to a few hydrophones and small aperture. Consequently, these arrays exhibit lower resolution of the estimated seabed properties, and the reliability of the environmental estimates may be questionable. The objective of the NATO-STO CMRE sea trial REP14-MED (conducted west of Sardinia, Mediterranean Sea) is to evaluate the performance of a prototype glider array with eight hydrophones in a line and variable hydrophone spacing for seabed characterization using natural-made ambient noise. This prototype array is deployed vertically above the seabed together with a 32-element reference vertical line array. The arrays are moored at different sites with varying sediment properties and stratification. The seabed reflection properties and layering structure at these sites are estimated from ambient noise using both arrays and the results are compared to assess the glider array performance. Synthetic extension of the glider array is performed to enhance resolution of the bottom properties, and the results are compared with these from the longer reference array.

2:45

2pAO5. Species classification of individual fish using the support vector machine. Atsushi Kinjo, Masanori Ito, Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai, MIyagi 981-3193, Japan, atsushi.kinjo@gmail.com), Tomohito Imaizumi, and Tomonari Akamatsu (Fisheries Res. Agency, National Res. Inst. of Fisheries Eng., Hasaki, Ibaraki, Japan)

The fish species classification using echo-sounder is important for fisheries. In the case of fish school of mixed species, it is necessary to classify individual fish species by isolating echoes from multiple fish. A broadband signal, which offered the advantage of high range resolution, was applied to detect individual fish for this purpose. The positions of fish were estimated

from the time difference of arrivals by using the split-beam system. The target strength (TS) spectrum of individual fish echo was computed from the isolated echo and the estimated position. In this paper, the Support Vector Machine was introduced to classify fish species by using these TS spectra. In addition, it is well known that the TS spectra are dependent on not only fish species but also fish size. Therefore, it is necessary to classify both fish species and size by using these features. We tried to classify two species and two sizes of schools. Subject species were chub mackerel (*Scomber japonicas*) and Japanese jack mackerel (*Trachurus japonicus*). We calculated the classification rates to limit the training data, frequency bandwidth and tilt angles. It was clarified that the best classification rate was 71.6 %. [This research was supported by JST, CREST.]

3:00–3:15 Break

3:15

2pAO6. Waveform inversion of ambient noise cross-correlation functions in a coastal ocean environment. Xiaoqin Zang, Michael G. Brown, Neil J. Williams (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149, xzang@rsmas.miami.edu), Oleg A. Godin (ESRL, NOAA, Boulder, CO), Nikolay A. Zabotin, and Liudmila Zabotina (CIRES, Univ. of Colorado, Boulder, CO)

Approximations to Green's functions have been obtained by cross-correlating concurrent records of ambient noise measured on near-bottom instruments at 5 km range in a 100 m deep coastal ocean environment. Inversion of the measured cross-correlation functions presents a challenge as neither ray nor modal arrivals are temporally resolved. We exploit both ray and modal expansion of the wavefield to address the inverse problem using two different parameterizations of the seafloor structure. The inverse problem is investigated by performing an exhaustive search over the relevant parameter space to minimize the integrated squared difference between computed and measured correlation function waveforms. To perform the waveform-based analysis described, it is important that subtle differences between correlation functions and Green's functions are accounted for. [Work supported by NSF and ONR.]

3:30

2pAO7. Application of time reversal to acoustic noise interferometry in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru), Oleg Godin (Univ. of Colorado, Boulder, CO), Jixing Qin (State Key Lab, Inst. of Acoust., Beijing, China), Nikolai Zabotin, Liudmila Zabotina (Univ. of Colorado, Boulder, CO), Michael Brown, and Neil Williams (Univ. of Miami, Miami, FL)

Two-point cross-correlations function (CCF) of diffuse acoustic noise approximates the Green's function, which describes deterministic sound propagation between the two measurement points. Similarity between CCFs and Green's functions motivates application to acoustic noise interferometry of the techniques that were originally developed for remote sensing using broadband, coherent compact sources. Here, time reversal is applied to CCFs of the ambient and shipping noise measured in 100 meter-deep water in the Straits of Florida. Noise was recorded continuously for about six days at three points near the seafloor by pairs of hydrophones separated by 5.0, 9.8, and 14.8 km. In numerical simulations, a strong focusing occurs in the vicinity of one hydrophone when the Green's function is back-propagated from the other hydrophone, with the position and strength of the focus being sensitive to density, sound speed, and attenuation coefficient in the bottom. Values of these parameters in the experiment are estimated by optimizing focusing of the back-propagated CCFs. The results are consistent with the values of the seafloor parameters evaluated independently by other means.

3:45

2pAO8. Shear wave inversion in a shallow coastal environment. Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Jennifer L. Giard (Marine Acoust., Inc., Middletown, RI), James H. Miller, Christopher D. P. Baxter (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Marcia J. Isakson, and Benjamin M. Goldsberry (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Estimation of the shear properties of seafloor sediments in littoral waters is important in modeling the acoustic propagation and predicting the strength of sediments for geotechnical applications. One of the promising approaches to estimate shear speed is by using the dispersion of seismo-acoustic interface (Scholte) waves that travel along the water-sediment boundary. 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 have been developed. The components of this inversion scheme are a genetic algorithm and a forward model which is based on dynamic stiffness matrix approach. The effects of the assumptions of the forward model on the inversion, particularly the shear wave depth profile, will be explored using a finite element model. The results obtained from a field test conducted in very shallow waters in Davisville, RI, will be presented. These results are compared to historic estimates of shear speed and recently acquired vibracore data. [Work sponsored by ONR, Ocean Acoustics.]

4:00

2pAO9. The effects of pH on acoustic transmission loss in an estuary. James H. Miller (Ocean Eng., Univ. of Rhode Island, URI Bay Campus, 215 South Ferry Rd., Narragansett, RI 02882, miller@egr.uri.edu), Laura Kloepper (Neurosci., Brown Univ., Providence, RI), Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Arthur J. Spivack, Steven D'Hondt, and Cathleen Turner (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI)

Increasing atmospheric CO_2 will cause the ocean to become more acidic with *p*H values predicted to be more than 0.3 units lower over the next 100 years. These lower pH values have the potential to reduce the absorption component of transmission loss associated with dissolved boron. Transmission loss effects have been well studied for deep water where *p*H is relatively stable over time-scales of many years. However, estuarine and coastal *p*H can vary daily or seasonally by about 1 *p*H unit and cause fluctuations in one-way acoustic transmission loss of 2 dB over a range of 10 km at frequencies of 1 kHz or higher. These absorption changes can affect the sound pressure levels received by animals due to identifiable sources such as impact pile driving. In addition, passive and active sonar performance in these estuarine and coastal waters can be affected by these *p*H fluctuations. Absorption changes in these shallow water environments offer a potential laboratory to study their effect on ambient noise due to distributed sources such as shipping and wind. We introduce an inversion technique based on perturbation methods to estimate the depth-dependent *p*H profile from measurements of normal mode attenuation. [Miller and Potty supported by ONR 322OA.]

Session 2pBA

Biomedical Acoustics: Quantitative Ultrasound II

Michael Oelze, Chair
UIUC, 405 N Mathews, Urbana, IL 61801

Contributed Papers

1:30

2pBA1. Receiver operating characteristic analysis for the detectability of malignant breast lesions in acousto-optic transmission ultrasound breast imaging. Jonathan R. Rosenfield (Dept. of Radiology, The Univ. of Chicago, 5316 South Dorchester Ave., Apt. 423, Chicago, IL 60615, jrosenfield@uchicago.edu), Jaswinder S. Sandhu (Santec Systems Inc., Arlington Heights, IL), and Patrick J. La Rivière (Dept. of Radiology, The Univ. of Chicago, Chicago, IL)

Conventional B-mode ultrasound imaging has proven to be a valuable supplement to x-ray mammography for the detection of malignant breast lesions in premenopausal women with high breast density. We have developed a high-resolution transmission ultrasound breast imaging system employing a novel acousto-optic (AO) liquid crystal detector to enable rapid acquisition of full-field breast ultrasound images during routine cancer screening. In this study, a receiver operating characteristic (ROC) analysis was performed to assess the diagnostic utility of our prototype system. Using a comprehensive system model, we simulated the AO transmission ultrasound images expected for a 1-mm malignant lesion contained within a dense breast consisting of 75% normal breast parenchyma and 25% fat tissue. A Gaussian noise model was assumed with SNRs ranging from 0 to 30. For each unique SNR, an ROC curve was constructed and the area under the curve (AUC) was computed to assess the lesion detectability of our system. For SNRs in excess of 10, the analysis revealed AUCs greater than 0.8983, thus demonstrating strong detectability. Our results indicate the potential for using an imaging system of this kind to improve breast cancer screening efforts by reducing the high false negative rate of mammography in premenopausal women.

1:45

2pBA2. 250-MHz quantitative acoustic microscopy for assessing human lymph-node microstructure. Daniel Rohrbach (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York City, NY 10038, drohrbach@RiversideResearch.org), Emi Saegusa-Beecroft (Dept. of General Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), Eugene Yanagihara (Kuakini Medical Ctr., Dept. of Pathol., Honolulu, HI), Junji Machi (Dept. of General Surgery, Univ. of Hawaii and Kuakini Medical Ctr., Honolulu, HI), Ernest J. Feleppa, and Jonathan Mamou (Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY)

We employed quantitative acoustic microscopy (QAM) to measure acoustic properties of tissue microstructure. 32 QAM datasets were acquired from 2, fresh and 11, deparaffinized, 12-μm-thick lymph-node samples obtained from cancer patients. Our custom-built acoustic microscope was equipped with an F-1.16, 250-MHz transducer having a 160-MHz bandwidth to acquire reflected signals from the tissue and a substrate that intimately contacted the tissue. QAM images with a spatial resolution of 7 μm were generated of attenuation (A), speed of sound (SOS), and acoustic impedance (Z). Samples then were stained using hematoxylin and eosin, imaged by light microscopy, and co-registered to QAM images. The spatial resolution and contrast of QAM images were sufficient to distinguish tissue regions consisting of lymphocytes, fat cells and fibrous tissue. Average properties for lymphocyte-dominated tissue were 1552.6 $\pm$ 30 m/s for SOS, 9.53 $\pm$ 3.6 dB/MHz/cm for A, and 1.58 $\pm$ 0.08 Mrayl for Z. Values for Z obtained from fresh samples agreed well with those obtained from 12-μm sections from the same node. Such 2D images provide a basis for developing improved ultrasound-scattering models underlying quantitative ultrasound methods currently used to detect cancerous regions within lymph nodes. [NIH Grant R21EB016117.]

2:00

2pBA3. Detection of sub-micron lipid droplets using transmission-mode attenuation measurements in emulsion phantoms and liver. Wayne Kreider, Ameen Tabatabai (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), and Yak-Nam Wang (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

In liver transplantation, donor liver viability is assessed by the both the amount and type of fat present in the organ. General guidelines dictate that livers with more than 30% fat should not be transplanted; however, a lack of available donor organs has led to the consideration of livers with more fat. As a part of this process, it is desirable to distinguish micro-vesicular fat (< 1 μm droplets) from macro-vesicular fat (~10 μm droplets). A method of evaluating the relative amounts of micro- and macro-fat is proposed based on transmission-mode ultrasound attenuation measurements. For an emulsion of one liquid in another, attenuation comprises both intrinsic losses in each medium and excess attenuation associated with interactions between media. Using an established coupled-phase model, the excess attenuation associated with a monodisperse population of lipid droplets was calculated with physical properties representative of both liver tissue and dairy products. Calculations predict that excess attenuation can exceed intrinsic attenuation and that a well-defined peak in excess attenuation at 1 MHz should occur for droplets around 0.8 μm in diameter. Such predictions are consistent with preliminary transmission-mode measurements in dairy products. [Work supported by NIH grants EB017857, EB007643, EB016118, and T32 DK007779.]

2:15

2pBA4. Using speckle statistics to improve attenuation estimates for cervical assessment. Viksit Kumar and Timothy Bigelow (Mech. Eng., Iowa State Univ., 4112 Lincoln Swing St., Unit 113, Ames, IA 50014, vkumar@iastate.edu)

Quantitative ultrasound parameters like attenuation can be used to observe microchanges in the cervix. To give a better estimate of attenuation we can use speckle properties to classify which attenuation estimates are valid and conform to the theory. For fully developed and only one type of scatterer, Rayleigh distribution models the signal envelope. But in tissues as the number of scatterer type increases and the speckle becomes unresolved Rayleigh model fails. Gamma distribution has been empirically shown to be the best fit among all the distributions. Since more than one scatterer type is present for our work we used a mixture of gamma distributions. EM algorithm was used to find the parameters of the mixture and on basis of that the tissue types were segmented from each other based on the criteria of different scattering properties. Attenuation estimates were then calculated for tissues of the same scattering type only. 67 Women underwent Transvaginal

scan and the attenuation estimates were calculated for them after segregation of tissues on scattering basis. Attenuation was seen to decrease as the time of delivery came closer.

2:30

2pBA5. Using two-dimensional impedance maps to study weak scattering in isotropic random media. Adam Luchies and Michael Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Matthews Ave, Urbana, IL 61801, luchies1@illinois.edu)

An impedance map (ZM) is a computational tool for studying weak scattering in soft tissues. Currently, three-dimensional (3D) ZMs are created from a series of adjacent histological slides that have been stained to emphasize acoustic scattering structures. The 3D power spectrum of the 3DZM may be related to quantitative ultrasound parameters such as the backscatter coefficient. However, constructing 3DZMs is expensive, both in terms of computational time and financial cost. Therefore, the objective of this study was to investigate using two-dimensional (2D) ZMs to estimate 3D power spectra. To estimate the 3D power spectrum using 2DZMs, the autocorrelation of 2DZMs extracted from a volume were estimated and averaged. This autocorrelation estimate was substituted into the 3D Fourier transform that assumes radial symmetry to estimate the 3D power spectrum. Simulations were conducted on sparse collections of spheres and ellipsoids to validate the proposed method. Using a single slice that intersected approximately 75 particles, a mean absolute error was achieved of 1.1 dB and 1.5 dB for sphere and ellipsoidal particles, respectively. The results from the simulations suggest that 2DZMs can provide accurate estimates of the power spectrum and are a feasible alternative to the 3DZM approach.

2:45

2pBA6. Backscatter coefficient estimation using tapers with gaps. Adam Luchies and Michael Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Matthews Ave., Urbana, IL 61801, luchies1@illinois.edu)

When using the backscatter coefficient (BSC) to estimate quantitative ultrasound (QUS) parameters such as the effective scatterer diameter (ESD) and the effective acoustic concentration (EAC), it is necessary to assume that the interrogated medium contains diffuse scatterers. Structures that invalidate this assumption can significantly affect the estimated BSC parameters in terms of increased bias and variance and decrease performance when classifying disease. In this work, a method was developed to mitigate the effects of non-diffuse echoes, while preserving as much signal as possible for obtaining diffuse scatterer property estimates. Specially designed tapers with gaps that avoid signal truncation were utilized for this purpose. Experiments from physical phantoms were used to evaluate the effectiveness of the proposed BSC estimation methods. The mean squared error (MSE) for BSC between measured and theoretical had an average value of approximately 1.0 and 0.2 when using a Hanning taper and PR taper, respectively, with six gaps. The BSC error due to amplitude bias was smallest for PR tapers with time-bandwidth product $N\omega = 1$. The BSC error due to shape bias was smallest for PR tapers with $N\omega = 4$. These results suggest using different taper types for estimating ESD versus EAC.

3:00

2pBA7. Application of the polydisperse structure function to the characterization of solid tumors in mice. Aiguo Han and William D. O'Brien (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801, han51@uiuc.edu)

A polydisperse structure function model has been developed for modeling ultrasonic scattering from dense scattering media. The polydisperse structure function is incorporated to a fluid-filled sphere scattering model to model the backscattering coefficient (BSC) of solid tumors in mice. Two types of tumors were studied: a mouse sarcoma (Englebreth-Holm-Swarm [EHS]) and a mouse carcinoma (4T1). The two kinds of tumors had significantly different microstructures. The carcinoma had a uniform distribution of carcinoma cells. The sarcoma had cells arranged in groups usually containing less than 20 cells per group, causing an increased scatterer size and size distribution. Excised tumors (13 EHS samples and 15 4T1 samples) were scanned using single-element transducers covering the frequency range 11–105 MHz. The BSC was estimated using a planar reference technique. The model was fit to the experimental BSC using a least-square fit. The mean scatterer radius and the Schulz width factor (which characterizes the width of the scatterer size distribution) were estimated. The results showed significantly higher scatterer size estimates and wider scatterer size distribution estimates for EHS than for 4T1, consistent with the observed difference in microstructure of the two types of tumors. [Work supported by NIH CA111289.]

3:15–3:30 Break

3:30

2pBA8. Experimental comparison of methods for measuring backscatter coefficient using single element transducers. Timothy Stiles and Andrew Selep (Phys., Monmouth College, 700 E Broadway Ave., Monmouth, IL 61462, tstiles@monmouthcollege.edu)

The backscatter coefficient (BSC) has promise as a diagnostic aid. However, measurements of the BSC of soft-tissue mimicking materials have proven difficult; results on the same samples by various laboratories have up to two orders of magnitude difference. This study compares methods of data analysis using data acquired from the same samples using single element transducers, with a frequency range of 1 to 20 MHz and pressure focusing gains between 5 and 60. The samples consist of various concentrations of milk in agar with scattering from glass microspheres. Each method utilizes a reference spectrum from a planar reflector but differ in the diffraction and attenuation correction algorithms. Results from four methods of diffraction correction and three methods of attenuation correction are compared to each other and to theoretical predictions. Diffraction correction varies from no correction to numerical integration of the beam throughout the data acquisition region. Attenuation correction varies from limited correction for the attenuation up to the start of the echo acquisition window, to correcting for attenuation within a numerical integration of the beam profile. Results indicate the best agreements with theory are the methods that utilize the numerical integration of the beam profile.

3:45

2pBA9. Numerical simulations of ultrasound-pulmonary capillary interaction. Brandon Patterson (Mech. Eng., Univ. of Michigan, 626 Spring St., Apt. #1, Ann Arbor, MI 48103-3200, awesome@umich.edu), Douglas L. Miller (Radiology, Univ. of Michigan, Ann Arbor, MI), David R. Dowling, and Eric Johnsen (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Although lung hemorrhage (LH) remains the only bioeffect of non-contrast, diagnostic ultrasound (DUS) proven to occur in mammals, a fundamental understanding of DUS-induced LH remains lacking. We hypothesize that the fragile capillary beds near the lungs surface may rupture as a result of ultrasound-induced strains and viscous stresses. We perform simulations of DUS waves propagating in tissue (modeled as water) and impinging on a planar lung surface (modeled as air) with hemispherical divots representing individual capillaries (modeled as water). Experimental ultrasound pulse waveforms of frequencies 1.5–7.5 MHz are used for the simulation. A high-order accurate discontinuity-capturing scheme solves the two-dimensional, compressible Navier-Stokes equations to obtain velocities, pressures, stresses, strains, and displacements in the entire domain. The mechanics of the capillaries are studied for a range of US frequencies and amplitudes. Preliminary results indicate a strong dependence of the total strain on the capillary size relative to the wavelength.

4:00

2pBA10. Acoustic radiation force due to nonaxisymmetric sound beams incident on spherical viscoelastic scatterers in tissue. Benjamin C. Treweek, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, btreweek@utexas.edu)

The theory for acoustic radiation force on a viscoelastic sphere of arbitrary size in tissue was extended recently to account for nonaxisymmetric incident fields [Ilinskii *et al.*, POMA **19**, 045004 (2013)]. A spherical harmonic expansion was used to describe the incident field. This work was specialized at the spring 2014 ASA meeting to focused axisymmetric sound

beams with various focal spot sizes and a scatterer located at the focus. The emphasis of the present contribution is nonaxisymmetric fields, either through moving the scatterer off the axis of an axisymmetric beam or through explicitly defining a nonaxisymmetric beam. This is accomplished via angular spectrum decomposition of the incident field, spherical wave expansions of the resulting plane waves about the center of the scatterer, Wigner D-matrix transformations to express these spherical waves in a coordinate system with the polar axis aligned with the desired radiation force component, and finally integration over solid angle to obtain spherical wave amplitudes as required in the theory. Various scatterer sizes and positions relative to the focus are considered, and the effects of changing properties of both the scatterer and the surrounding tissue are examined. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:15

2pBA11. Convergence of Green's function-based shear wave simulations in models of elastic and viscoelastic soft tissue. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., Michigan State University, East Lansing, MI, yiqunyang.nju@gmail.com), Matthew Urban (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN), and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Green's functions effectively simulate shear waves produced by an applied acoustic radiation force in elastic and viscoelastic soft tissue. In an effort to determine the optimal parameters for these simulations, the convergence of Green's function-based calculations is evaluated for realistic spatial distributions of the initial radiation force "push." The input to these calculations is generated by FOCUS, the "Fast Object-oriented C++ Ultrasound Simulator," which computes the approximate intensity fields generated by a Phillips L7-4 ultrasound transducer array for both focused and unfocused beams. The radiation force in the simulation model, which is proportional to the simulated intensity, is applied for 200 μs, and the resulting displacements are calculated with the Green's function model. Simulation results indicate that, for elastic media, convergence is achieved when the intensity field is sampled at roughly one-tenth of the wavelength of the compressional component that delivers the radiation force "push." Aliasing and oscillation artifacts are observed in the model for an elastic medium at lower sampling rates. For viscoelastic media, spatial sampling rates as low as two samples per compressional wavelength are sufficient due to the low-pass filtering effects of the viscoelastic medium. [Supported in part by NIH Grants R01EB012079 and R01DK092255.]

4:30

2pBA12. Quantifying mechanical heterogeneity of breast tumors using quantitative ultrasound elastography. Tengxiao Liu (Dept. of Mech., Aerosp. and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY), Olalekan A. Babaniyi (Mech. Eng., Boston Univ., Boston, MA), Timothy J. Hall (Medical Phys., Univ. of Wisconsin, Wisconsin, WI), Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), and Assad A. Oberai (Dept. of Mech., Aerosp. and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY)

Heterogeneity is a hallmark of cancer whether one considers the genotype of cancerous cells, the composition of their microenvironment, the distribution of blood and lymphatic microvasculature, or the spatial distribution of the desmoplastic reaction. It is logical to expect that this heterogeneity in tumor microenvironment will lead to spatial heterogeneity in its mechanical properties. In this study we seek to quantify the mechanical heterogeneity within malignant and benign tumors using ultrasound based elasticity imaging. By creating *in-vivo* elastic modulus images for ten human subjects with breast tumors, we show that Young's modulus distribution in cancerous breast tumors is more heterogeneous when compared with tumors that are not malignant, and that this signature may be used to distinguish malignant breast tumors. Our results complement the view of cancer as a heterogeneous disease by demonstrating that mechanical properties within cancerous tumors are also spatially heterogeneous. [Work supported by NIH, NSF.]

4:45

2pBA13. Convergent field elastography. Michael D. Gray, James S. Martin, and Peter H. Rogers (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405, michael.gray@me.gatech.edu)

An ultrasound-based system for non-invasive estimation of soft tissue shear modulus will be presented. The system uses a nested pair of transducers to provide force generation and motion measurement capabilities. The outer annular element produces a ring-like ultrasonic pressure field distribution. This in turn produces a ring-like force distribution in soft tissue, whose response is primarily observable as a shear wave field. A second ultrasonic transducer nested inside the annular element monitors the portion of the shear field that converges to the center of the force distribution pattern. Propagation speed is estimated from shear displacement phase changes resulting from dilation of the forcing radius. Forcing beams are modulated in order to establish shear speed frequency dependence. Prototype system data will be presented for depths of 10–14 cm in a tissue phantom, using drive parameters within diagnostic ultrasound safety limits. [Work supported by ONR and the Neely Chair in Mechanical Engineering, Georgia Institute of Technology.]

5:00

2pBA14. Differentiation of benign and malignant breast lesions using Comb-Push Ultrasound Shear Elastography. Max Denis, Mohammad Mehmohammadi (Physiol. and Biomedical Eng., Mayo Clinic, 200 First St. SW, Rochester, MN 55905, denis.max@mayo.edu), Duane Meixner, Robert Fazzio (Radiology-Diagnostic, Mayo Clinic, Rochester, MN), Shigao Chen, Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN), and Azra Alizad (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, Missouri)

In this work, the results from our Comb Push Ultrasound Shear Elastography (CUSE) assessment of suspicious breast lesions are presented. The elasticity value of the breast lesions are correlated to histopathological findings to evaluate their diagnostic value in differentiating between malignant and benign breast lesions. A total of 44 patients diagnosed with suspicious breast lesions were evaluated using CUSE prior to biopsy. All patient study procedures were conducted according to the protocol approved by Mayo Clinic Institutional Review Board (IRB). Our cohort consisted of 27 malignant and 17 benign breast lesions. The results indicate an increase in shear wave velocity in both benign and malignant lesions compared to normal breast tissue. Furthermore, the Young's modulus is significantly higher in malignant lesions. An optimal cut-off value of the Young's modulus $\geq$ 80 kPa was observed for the receiver operating characteristic (ROC) curve. This is concordant with the published cut-off values of elasticity for suspicious breast lesions. [This work is supported in part by the grant 3R01CA148994-04S1 and 5R01CA148994-04 from NIH.]

5:15

2pBA15. Comparison between diffuse infrared and acoustic transmission over the human skull. Qi Wang, Namratha Reganti, Yutoku Yoshioka, Mark Howell, and Gregory T. Clement (BME, LRI, Cleveland Clinic, 9500 Euclid Ave., Cleveland, OH 44195, qiqiwang83@gmail.com)

Skull-induced distortion and attenuation present a challenge to both transcranial imaging and therapy. Whereas therapeutic procedures have been successful in offsetting aberration using from prior CTs, this approach impractical for imaging. In effort to provide a simplified means for aberration correction, we have been investigating the use of diffuse infrared light as an indicator of acoustic properties. Infrared wavelengths were specifically selected for tissue penetration; however this preliminary study was performed through bone alone via a transmission mode to facilitate comparison with acoustic measurements. The inner surface of a half human skull, cut along the sagittal midline, was illuminated using an infrared heat lamp and images of the outer surface were acquired with an IR-sensitive camera. A range of source angles were acquired and averaged to eliminate source bias. Acoustic measurement were likewise obtained over the surface with a source (1 MHz, 12.7 mm-diam) oriented parallel to the skull surface and hydrophone receiver (1 mm PVDF). Preliminary results reveal a positive correlation between sound speed and optical intensity, whereas poor correlation is observed between acoustic amplitude and optical intensity. [Work funded under NIH R01EB014296.]

5:30

2pBA16. A computerized tomography system for transcranial ultrasound imaging. Sai Chun Tang (Dept. of Radiology, Harvard Med. School, 221 Longwood Ave., Rm. 521, Boston, MA 02115, sct@bwh.harvard.edu) and Gregory T. Clement (Dept. of Biomedical Eng., Cleveland Clinic, Cleveland, OH)

Hardware for tomographic imaging presents both challenge and opportunity for simplification when compared with traditional pulse-echo imaging systems. Specifically, point diffraction tomography does not require simultaneous powering of elements, in theory allowing just a single transmit channel and a single receive channel to be coupled with a switching or multiplexing network. In our ongoing work on transcranial imaging, we have developed a 512-channel system designed to transmit and/or receive a high voltage signal from/to arbitrary elements of an imaging array. The overall design follows a hierarchy of modules including a software interface, microcontroller, pulse generator, pulse amplifier, high-voltage power converter, switching mother board, switching daughter board, receiver amplifier, analog-to-digital converter, peak detector, memory, and USB communication. Two pulse amplifiers are included, each capable producing up to 400 Vpp via power MOSFETS. Switching is based around mechanical relays that allow passage of 200 V, while still achieving switching times of under 2 ms, with an operating frequency ranging from below 100 kHz to 10 MHz. The system is demonstrated through ex vivo human skulls using 1 MHz transducers. The overall system design is applicable to planned human studies in transcranial image acquisition, and may have additional tomographic applications for other materials necessitating a high signal output. [Work was supported by NIH R01 EB014296.]

TUESDAY AFTERNOON, 28 OCTOBER 2014 — INDIANA C/D, 2:45 P.M. TO 3:30 P.M.

Session 2pEDa

Education in Acoustics: General Topics in Education in Acoustics

Uwe J. Hansen, Chair

Chemistry& Physics, Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

Contributed Papers

2:45

2pEDa1. @acousticsorg: The launch of the Acoustics Today twitter feed. Laura N. Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laura_kloepper@brown.edu) and Daniel Farrell (Web Development office, Acoust. Society of America, Melville, NY)

Acoustics Today has recently launched our twitter feed, @acousticsorg. Come learn how we plan to spread the mission of Acoustics Today, promote the science of acoustics, and connect with acousticians worldwide! We will also discuss proposed upcoming social media initiatives and how you, an ASA member, can help contribute. This presentation will include an extended question period in order to gather feedback on how Acoustics Today can become more involved with social media.

3:00

2pEDa2. Using Twitter for teaching. William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, wvslaton@uca.edu)

The social media microblogging platform, Twitter, is an ideal avenue to learn about new science in the field of acoustics as well as to share that new-found information with students. As a user discovers a network of science bloggers and journalists to follow the amount of science uncovered grows. Conversations between science writers and scientists themselves enhance this learning opportunity. Several examples of using twitter for teaching will be presented.

3:15

2pEDa3. Unconventional opportunities to recruit future science, technology, engineering, and math scholars. Roger M. Logan (Teledyne, 12338 Westella, Houston, TX 77077, rogermlogan@sbcglobal.net)

Pop culture conventions provide interesting and unique opportunities to inspire the next generation of STEM contributors. Literary, comic, and anime are a few example of this type of event. This presentation will provide insights into these venues as well as how to get involved and help communicate that careers in STEM can be fun and rewarding.

Session 2pEDb

Education in Acoustics: Take 5's

Uwe J. Hansen, Chair
Chemistry& Physics, Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign-up for two consecutive slots.

TUESDAY AFTERNOON, 28 OCTOBER 2014 INDIANA E, 1:55 P.M. TO 5:00 P.M.

Session 2pID

Interdisciplinary: Centennial Tribute to Leo Beranek's Contributions in Acoustics

William J. Cavanaugh, Cochair
Cavanaugh Tocci Assoc. Inc., 3 Merifield Ln., Natick, MA 01760-5520

Carl Rosenberg, Cochair
Acentech Incorporated, 33 Moulton Street, Cambridge, MA 02138

Chair's Introduction—1:55

Invited Papers

2:00

2pID1. Leo Beranek's role in the Acoustical Society of America. Charles E. Schmid (10677 Manitou Pk. Blvd., Bainbridge Island, WA 98110, cechmid@att.net)

Leo Beranek received the first 75th anniversary certificate issued by the Acoustical Society of America commemorating his long-time association with the Society at the joint ASA/ICA meeting in Montreal in 2013. Both the Society and Leo have derived mutual benefits from this long and fruitful association. Leo has held many important roles as leader in the ASA. He served as vice president (1949–1950), president (1954–1955), Chair of the Z24 Standards Committee (1950–1953), meeting organizer (he was an integral part of the Society's 25th, 50th, and 75th Anniversary meetings), associate editor (1950–1959), author of three books sold via ASA, publisher of 75 peer-reviewed JASA papers, and presented countless papers at ASA meetings. Much of his work has been recognized by the Society which presented him with the R. Bruce Lindsay Award (1944), the Wallace Clement Sabine Award (1961), the Gold Medal (1975), and an Honorary Fellowship (1994). He has participated in the Acoustical Society Foundation and donated generously to it. He has been an inspiration for younger Society members (which include all of us on this occasion celebrating his 100th birthday).

2:15

2pID2. Leo Beranek's contributions to noise control. George C. Maling (INCE FOUNDATION, 60 High Head Rd., Harpswell, ME 04079, INCEUSA@aol.com) and William W. Lang (INCE FOUNDATION, Poughkeepsie, New York)

Leo Beranek has made contributions to noise control for many years, beginning with projects during World War II when he was a Harvard University. Later, at MIT, he taught a course (6.35) which included noise control, and ran MIT summer courses on the subject. His book, *Noise Reduction*, was published during that time. Additional books followed. Noise control became an important part of the consulting work at Bolt Beranek and Newman. Two projects are of particular interest: The efforts to silence a wind tunnel in Cleveland,

Ohio, and the differences in noise emissions and perception as the country entered the jet age. Leo was one of the founders of the Institute of Noise Control Engineering, and served as its charter president. Much of the success of the Institute is due to his early leadership. He has also played an important role in noise policy, beginning in the late 1960s and, in particular, with the passage of the Noise Control Act of 1972. This work continued into the 1990s with the formation of the "Peabody Group," and cooperation with the National Academy of Engineering in the formation of noise policy.

2:30

2pID3. Beranek's porous material model: Inspiration for advanced material analysis and design. Cameron J. Fackler and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, facklc@rpi.edu)

In 1942, Leo Beranek presented a model for predicting the acoustic properties of porous materials [J. Acoust. Soc. Am. 13, 248 (1942)]. Since then, research into many types of porous materials has grown into a broad field. In addition to Beranek's model, many other models for predicting the acoustic properties of porous materials in terms of key physical material parameters have been developed. Following a brief historical review, this work concentrates on studying porous materials and microperforated panels—pioneered by one of Beranek's early friends and fellow students, Dah-You Maa. Utilizing equivalent fluid models, porous material and microperforated panel theories have recently been unified. In this work, the Bayesian inference framework is applied to single- and multilayered porous and microperforated materials. Bayesian model selection and parameter estimation are used to guide the analysis and design of innovative multilayer acoustic absorbers.

2:45

2pID4. Technology, business, and civic visionary. David Walden (retired from BBN, 12 Linden Rd., East Sandwich 02537, dave@walden-family.com)

In high school and college, Leo Beranek was already developing the traits of an entrepreneur. At Bolt Beranek and Newman he built a culture of innovation. He and his co-founders also pursued a policy of looking for financial returns, via diversification and exploitation of intellectual property, beyond their initial acoustics-based professional services business. In particular, in 1956–1957 Leo recruited J.C.R. Licklider to help BBN move into the domain of computers. In time, information sciences and computing became as significant a business for BBN as acoustics. While BBN did innovative work in many areas of computing, perhaps the most visible area was with the technology that became the Internet. In 1969, Leo left day-to-day management of BBN, although he remained associated with the company for more years. Beyond BBN, Leo worked, often in a leadership role, with a variety of institutions to improve civic life and culture around Boston.

3:00

2pID5. Leo Beranek and concert hall acoustics. Benjamin Markham (Acentech Inc, 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Dr. Leo Beranek's pioneering concert hall research and project work has left an indelible impression on the study and practice of concert hall design. Working as both scientist and practitioner simultaneously for most of his 60+ years in the industry, his accomplishments include dozens of published papers on concert hall acoustics, several seminal books on the subject, and consulting credit for numerous important performance spaces. This paper will briefly outline a few of his key contributions to the field of concert hall acoustics (including his work regarding audience absorption, the loudness parameter G, the system of concert hall metrics and ratings that he developed, and other contributions), his project work (including the Tanglewood shed, Philharmonic Hall, Tokyo Opera City concert hall, and others), and his role as an inspiration for other leaders in the field. His work serves as the basis, the framework, the inspiration, or the jumping-off point for a great deal of current concert hall research, as evidenced by the extraordinarily high frequency with which his work is cited; this paper will conclude with some brief remarks on the future of concert hall research that will build on Dr. Beranek's extraordinary career.

3:15

2pID6. Concert hall acoustics: Recent research. Leo L. Beranek (Retired, 10 Longwood Dr., Westwood, MA 02090, beranekleo@ieee.org)

Recent research on concert hall acoustics is reviewed. Discussed are (1) ten top quality halls acoustically; (2) listeners acoustical preferences; (3) how musical dynamics are enhanced by hall shape; (4) effect of seat upholstering on sound strength and hall dimensions; (5) recommended minimum and maximum hall dimensions and audience capacities in shoebox, surround, and fan shaped halls.

3:30

2pID7. Themes of thoughts and thoughtfulness. Carl Rosenberg (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, crosenberg@acentech.com) and William J. Cavanaugh (Cavanaugh/Tocci, Sudbury, MA)

In preparing and compiling the background for the issue of Acoustics Today on Leo Beranek to commemorate his 100th birthday, there were some consistent themes of Leo's work and contribution to colleagues and scholars with whom he worked. This was particularly evident in the many "side-bars" solicited from over three dozen friends and colleagues. The authors discuss these patterns and share insights on the manner in which Leo was most influential. There will be opportunities for audience participants to share their thoughts and birthday greetings with Leo.

3:45–4:15 Panel Discussion

4:15–5:00 Celebration

Session 2pMU

Musical Acoustics: Synchronization Models in Musical Acoustics and Psychology

Rolf Bader, Chair
Institute of Musicology, University of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany

Invited Papers

1:00

2pMU1. Models and findings of synchronization in musical acoustics and music psychology. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

Synchronization is a crucial mechanism in music tone production and perception. With wind instruments, the overtone series of notes synchronize to nearly perfect harmonic relations due to nonlinear effects and turbulence at the driving mechanism although the overblown pitches of flutes or horns may differ considerably from such a simple harmonic relation. Organ pipes close to each other synchronize in pitch by interaction of the sound pressures. With violins, the sawtooth motion appears because of a synchronization of the stick/slip interaction with the string length. All these models are complex systems also showing bifurcations in terms of multiphonics, biphonation or subharmonics. On the subjects perception and music production side models of synchronization, like the free-energy principle modeling perception by minimizing surprise and adaptation to physical parameters of sound production, neural nets of timbre, tone, or rhythm perception or synergetic models of rhythm production are generally suited much better to model music perception than simplified linear models.

2p TUE. PM

1:20

2pMU2. One glottal airflow—Two vocal folds. 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) and Ingo R. Titze (Dept. of Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA)

Vocalization for speech and singing involves self-sustained oscillation between a stream of air and a pair of vocal folds. Each vocal fold has its own set of natural frequencies (modes of vibration) governed by the viscoelastic properties of tissue layers and their boundary conditions. Due to asymmetry, the modes of left and right vocal folds are not always synchronized. The common airflow between them can entrain the modes, but not always in a 1:1 ratio. Examples of bifurcations are given for human and animal vocalization, as well as from computer simulation. Vocal artists may use desynchronization for special vocal effects. Surgeons who repair vocal folds make decisions about the probability of regaining synchronization when one vocal fold is injured. Complete desynchronization, allowing only one vocal fold to oscillate, may be a better strategy in some cases then attempting to achieve symmetry.

1:40

2pMU3. Synchronization of organ pipes—Experimental facts and theory. Markus W. Abel and Jost L. Fischer (Inst. for Phys. and AstroPhys., Potsdam Univ., Karl/Liebknecht Str. 24-25, Potsdam 14469, Germany, markus.abel@physik.uni-potsdam.de)

Synchronization of musical instruments has raised attention due to the important implications on sound production in musical instruments and technological applications. In this contribution, we show new results on the interaction of two coupled organ pipes: we present a new experiment where the pipes were positioned in a plane with varying distance, further we briefly refer to a corresponding description in terms of a reduced model, and eventually show numerical simulations which are in full agreement with the measurements. Experimentally, the 2D setup allows for the observation of a new phenomenon: a synchronization/desynchronization transition at regular distances of the pipes. The developed model basically consists of a self-sustained oscillator with nonlinear, delayed coupling. The nonlinearity reflects the complicated interaction of emitted acoustical waves with the jet exiting at the organ pipe mouth, and the delay term takes care of the wave propagation. Synchronization is a clear evidence for the importance of nonlinearities in music and continues to be a source of astonishing results.

2:00

2pMU4. Nonlinear coupling mechanisms in acoustic oscillator systems which can lead to synchronization. Jost Fischer (Dept. for Phys. and Astronomy, Univ. of Potsdam, Karl-Liebknecht-Str 24/25, Potsdam, Brandenburg 14476, Germany, jost.fischer@uni-potsdam.de) and Markus Abel (Ambrosys GmbH, Potsdam, Germany)

We present results on the coupling mechanisms in wind-driven, self-sustained acoustic oscillators. Such systems are found in engineering applications, as gas burners, and—more beautiful—in musical instruments. As a result, we find that coupling and oscillators are nonlinear in character, which can lead to synchronization. We demonstrate our ideas using one of the oldest and most complex musical devices: organ pipes. Building up on the questions of preceding works, the elements of the sound generation are identified using detailed

experimental and theoretical studies, as well as numerical simulations. From these results, we derive the nonlinear coupling mechanisms of the mutual interaction of organ pipes. This leads to a nonlinear coupled acoustic oscillator model, which is based on the aeroacoustical and fluid dynamical first principles. The model calculations are compared with the experimental results from preceding works. It appears that the sound generation and the coupling mechanisms are properly described by the developed nonlinear coupled model of self-sustained oscillators. In particular, we can explain the unusual nonlinear shape Arnold tongues of the coupled two-pipe system. Finally, we show the power of modern CFD simulations by a 2D simulation of two mutually interacting organ pipes, i.e., the compressible Navier-Stokes equations are numerically solved.

2:20–2:40 Break

2:40

2pMU5. Auditory-inspired pitch extraction using a synchrony capture filterbank. Kumaresan Ramdas, Vijay Kumar Peddinti (Dept. of Elec., Comput. Eng., Univ. of Rhode Island, Kelley A216 4 East Alumni Ave., Kingston, RI 02881, kumar@ele.uri.edu), and Peter Cariani (Hearing Res. Ctr. & Dept. of Biomedical Eng., Boston Univ., Boston, MA)

The question of how harmonic sounds in speech and music produce strong, low pitches at their fundamental frequencies, F0's, has been of theoretical and practical interest to scientists and engineers for many decades. Currently the best auditory models for F0 pitch, (e.g., Meddis & Hewitt, 1991), are based on bandpass filtering (cochlear mechanics), half-wave rectification and low-pass filtering (hair cell transduction, synaptic transmission), channel autocorrelations (all-order interspike interval distributions) aggregated into a summary autocorrelation, followed by an analysis that determines the most prevalent interspike intervals. As a possible alternative to explicit autocorrelation computations, we propose an alternative model that uses an adaptive Synchrony Capture Filterbank (SCFB) in which channels in a filterbank neighborhood are driven exclusively (captured) by dominant frequency components closest to them. Channel outputs are then adaptively phase aligned with respect to a common time reference to compute a Summary Phase Aligned Function (SPAF), aggregated across all channels, from which F0 can then be easily extracted. Possible relations to brain rhythms and phase-locked loops are discussed. [Work supported by AFSOR FA9550-09-1-0119, Invited to special session about Synchronization in Musical Acoustics and Music Psychology.]

3:00

2pMU6. Identification of sound sources in soundscape using acoustic, psychoacoustic, and music parameters. Ming Yang and Jian Kang (School of Architecture, Univ. of Sheffield, Western Bank, Sheffield S10 2TN, United Kingdom, arp08my@sheffield.ac.uk)

This paper explores the possibility of automatic identification/classification of environmental sounds, by analyzing sound with a number of acoustic, psychoacoustic, and music parameters, including loudness, pitch, timbre, and rhythm. The sound recordings of single sound sources labeled in four categories, i.e., water, wind, birdsongs, and urban sounds including street music, mechanical sounds and traffic noise, are automatically identified with machine learning and mathematic models, including artificial neural networks and discriminant functions, based on the results of the psychoacoustic/music measures. The accuracies of the identification are above 90% for all the four categories. Moreover, based on these techniques, identification of construction noise sources from general urban background noise is explored, using the large construction project of London Bridge Station redevelopment as a case study site.

Contributed Papers

3:20

2pMU7. Neuronal synchronization of musical large-scale form. Lenz Hartmann (Insitut for Systematic Musicology, Universität Hamburg, Feldstrasse 59, Hamburg, Hamburg 20357, Germany, lenz.hartmann@gmx.de)

Musical form in this study is taken as structural aspects of music ranging over several bars as a combination of all elements that constitutes a piece of music, like pitch, rhythm or timbre. In an EEG-study 25 participants listen to the first about four minutes of a piece of electronic dance music for three times each and ERP grand-averages were calculated. Correlations of a one second time windows between the ERPs of all electrodes and therefore of different brain regions is used as a measure of synchronization between these areas. Local maxima corresponding to strong synchronization show up at expectancy points of boundaries in the present musical form. A modified FFT analysis of the ERPs of the mean of all trials and all channels that just take the ten highest peaks in consideration show strongest brain activity at frequencies in the gamma-band (about 40–60 Hz) and in the beta-band (about 20–30 Hz).

3:35

2pMU8. Nonlinearities and self-organization in the sound production of the Rhodes piano. Malte Muenster, Florian Pfeifle, Till Weinrich, and Martin Keil (Systematic Musicologie, Univ. of Hamburg, Pilatuspool, 19, Hamburg, Hamburg 20355, Germany, m.muenster@arcor.de)

Over the last five decades the Rhodes piano became a common keyboard instrument. It is played in such diverse musical genres as Jazz, Funk, Fusion, or Pop. The sound processing of the Rhodes has not been studied in detail beforeIts sound is produced by a mechanical driven tuning fork like system causing a change in the magnetic flux of an electromagnetic pick up system. The mechanical part of the tone production consists of a small diameter tine made of stiff spring steel, the tine, and a tone bar made of brass, which is strongly coupled to the former and acts as a resonator. The system is an example for strong generator-resonator coupling. The tine acts as a generator forcing the tonebar to vibrate with its fundamental frequency. Despite of extremely different and much lower eigenfrequencies the tonebar is enslaved by the tine. The tine is of lower spatial dimension and less damped and acts nearly linear. The geometry of the tonebar is much more complex and therefore of higher dimension and damped stronger. The vibration of these two parts are perfectly in-phase or anti-phase pointing to a quasi-synchronization behavior. Moreover, the tonebar is responsible for the timbre of the initial transient. It adds the glockenspiel sound to the transient and extends the sustain. The sound production is discussed as synergetic, self-organizing system, leading to a very precise harmonic overtone structure and characteristic initial transients enhancing the variety of musical performance.

Session 2pNSa

Noise and Psychological and Physiological Acoustics: New Frontiers in Hearing Protection II

Elliott H. Berger, Cochair
Occupational Health & Environmental Safety Division, 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

William J. Murphy, Cochair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998

Chair's Introduction—1:25

Invited Paper

1:30

2pNSa1. Comparison of impulse peak insertion loss measured with gunshot and shock tube noise sources. 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), Elliott H. Berger (Personal Safety Div., E-A-RCAL lab, 3M, Indianapolis, IN), and William A. Ahroon (US Army Aeromedical Res. Lab., US Army, Fort Rucker, AL)

The National Institute for Occupational Safety and Health in cooperation with scientists from 3M and the U.S. Army Aeromedical Research Laboratory conducted a series of Impulse peak insertion loss (IPIL) tests of the acoustic test fixtures from the Institute de Saint Louis (ISL) with a 0.223 caliber rifle and two different acoustic shock tubes. The Etymotic Research ETYPlugs™ earplug, 3M™ TactiCalPro™ communication headset and the dual protector combination were tested with all three impulse noise sources. The spectra, IPIL, and the reduction of different damage risk criteria will be presented. The spectra from the noise sources vary considerably with the rifle having peak energy at about 1000 Hz. The shock tubes had peak levels around 125 and 250 Hz. The IPIL values for the rifle were greater than those measured with the two shock tubes. The shock tubes had comparable IPIL results except at 150 dB for the dual protector condition. The treatment of the double protection condition is complicated because the earmuff reduces the shock wave and reduces the effective level experienced by the earplug. For the double protection conditions, bone conduction presents a potential limiting factor for the effective attenuation that can be achieved by hearing protection.

Contributed Paper

1:50

2pNSa2. Evaluation of level-dependent performance of in-ear hearing protection devices using an enclosed sound source. Theodore F. Argo and G. Douglas Meegan (Appl. Res. Assoc., Inc., 7921 Shaffer Parkway, Littleton, CO 80127, targo@ara.com)

Hearing protection devices are increasingly designed with the capability to protect against impulsive sound. Current methods used to test protection from impulsive noise, such as blasts and gunshots, suffer from various drawbacks and complex, manual experimental procedures. For example, the use of a shock tube to emulate blast waves typically produces a blast wind of a much higher magnitude than that generated by an explosive, a specific but important inconsistency between the test conditions and final application. Shock tube test procedures are also very inflexible and provide only minimal insight into the function and performance of advanced electronic hearing protection devices that may have relatively complex response as a function of amplitude and frequency content. To address the issue of measuring the amplitude-dependent attenuation provided by a hearing protection device, a method using a compression driver attached to an enclosed waveguide was developed. The hearing protection device is placed at the end of the waveguide and the response to exposure to impulsive and frequency-dependent signals at calibrated levels is measured. Comparisons to shock tube and standard frequency response measurements will be discussed.

Invited Papers

2:05

2pNSa3. Exploration of flat hearing protector attenuation and sound detection in noise. Christian Giguere (Audiology/SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H8M5, Canada, cgiguere@uottawa.ca) and Elliott H. Berger (Personal Safety Div., 3M, Indianapolis, IN)

Flat-response devices are a class of hearing protectors with nearly uniform attenuation across frequency. These devices can protect the individual wearer while maintaining the spectral balance of the surrounding sounds. This is typically achieved by reducing the muffling effect of conventional hearing protectors which provide larger attenuation at higher than lower frequencies, especially with earmuffs. Flat hearing protectors are often recommended when good speech communication or sound perception is essential, especially for wearers with high-frequency hearing loss, to maintain audibility at all frequencies. However, while flat-response devices are described in some acoustical standards, the tolerance limits for the definition of flatness are largely unspecified and relatively little is known on the exact conditions when such devices can be beneficial. The purpose of this study is to gain insight into the interaction between the spectrum of the noise, the shape of the attenuation-frequency response, and the hearing loss configuration on detection thresholds using a psychoacoustic model of sound detection in noise.

2:25

2pNSa4. Electronic sound transmission hearing protectors and horizontal localization: Training/adaptation effects. John Casali (Auditory Systems Lab, Virginia Tech, 250 Durham Hall, Blacksburg, VA 24061, jcasali@vt.edu) and Martin Robinette (U.S. Army Public Health Command, U.S. Army, Aberdeen Proving Ground, MD)

Auditory situation awareness is known to be affected by some hearing protectors, even advanced electronic devices. A horizontal localization task was employed to determine how use/training with electronic sound transmission hearing protectors affected auditory localization ability, as compared to open-ear. Twelve normal-hearing participants performed baseline localization testing in a hemi-anechoic field in three listening conditions: open-ear, in-the-ear (ITE) device (Etymotic EB-15), and over-the-ear (OTE) device (Peltor ComTac II). Participants then wore either the ITE or OTE protector for 12, almost daily, one-hour training sessions. Post-training, participants again underwent localization testing with all three conditions. A computerized custom software-hardware interface presented localization sounds and collected accuracy and timing measures. ANOVA and post hoc statistical tests revealed that pre-training localization performance with either the ITE or OTE protector was significantly worse ($p<0.05$) than open-ear performance. After training with any given listening condition, performance in that condition improved, in part from a practice effect. However, post-training localization showed near equal performance between the open-ear and the protector on which training occurred. Auditory learning, manifested as significant localization accuracy improvement, occurred for the training device, but not for the non-training device, i.e., no crossover benefit from the training device to the non-training device occurred.

Contributed Papers

2:45

2pNSa5. Measuring effective detection and localization performance of hearing protection devices. Richard L. McKinley (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, richard.mckinley.1@us.af.mil), Eric R. Thompson (Ball Aerosp. and Technologies, Air Force Res. Lab., Wright-Patterson AFB, OH), and Brian D. Simpson (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Awareness of the surrounding acoustic environment is essential to the safety of persons. However, the use of hearing protection devices can degrade the ability to detect and localize sounds, particularly quiet sounds. There are ANSI/ASA standards describing methods for measuring attenuation, insertion loss, and speech intelligibility in noise for hearing protection devices, but currently there are no standard methods to measure the effects of hearing protection devices on localization and/or detection performance. A method for measuring the impact of hearing protectors on effective detection and localization performance has been developed at AFRL. This method measures the response time in an aurally aided visual search task where the presentation levels are varied. The performance with several level-dependent hearing protection devices will be presented.

3:00

2pNSa6. Personal alert safety system localization field tests with firefighters. Joelle I. Suits, Casey M. Farmer, Ofodike A. Ezekoye (Dept. of Mech. Eng., The Univ. of Texas at Texas, 204 E Dean Keaton St., Austin, TX 78712, jsuits@utexas.edu), Mustafa Z. Abbasi, and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

When firefighters get lost or incapacitated on the fireground, there is little time to find them. This project has focused on a contemporary device used in this situation, the Personal Alert Safety System. We have studied the noises on the fireground (i.e., chainsaws, gas powered ventilation fans, pumper trucks) [J. Acoust. Soc. Am. **134**, 4221 (2013)], how the fire environment affects sound propagation [J. Acoust. Soc. Am. **134**, 4218 (2013)], and how firefighter personal protective equipment (PPE) affects human hearing [POMA **19**, 030054 (2013)]. To put all these pieces together, we have traveled to several fire departments across the country conducting tests to investigate how certain effects manifest themselves when firefighters search for the source of a sound. We tasked firefighters to locate a target sound in various acoustic environments while their vision was obstructed and while wearing firefighting PPE. We recorded how long it took them to find the source, what path they took, when they first heard the target sound, and the frequency content and sound pressure level of the acoustic environment. The results will be presented in this talk. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

3:15

2pNSa7. Noise level from burning articles on the fireground. Mustafa Z. Abbasi, Preston S. Wilson (Appl. Res Lab and Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78751, mustafa_abbasi@utexas.edu), and Ofodike A. Ezekoye (Dept. of Mech. of Eng., The Univ. of Texas at Austin, Austin, TX)

Firefighters encounter an extremely difficult environment due to the presence of heat, smoke, falling debris etc. If one of them needs rescue, an audible alarm is used to alert others of their location. This alarm, known as the Personal Alert Safety System (PASS) alarm, has been part of firefighter gear since the early 1980s. The PASS has been enormously successful, but a review of The National Institute for Occupational Safety and Health (NIOSH) firefighter fatality report suggests that there are instances when the alarm is not heard or not localized. In the past, we have studied fireground noise from various pieces of gear such as chainsaws and fans, etc., to understand the soundscape present during a firefighting operation. However, firefighters, and other interested parties have raised the issue of noise caused by the fire itself. The literature shows that buoyancy-controlled, non-premixed flames aerodynamically oscillate in the 10–16 Hz range, depending on the diameter of the fuel base. Surprisingly, few acoustic measurements have been made even for these relatively clean fire conditions. However, experience suggests burning items do create sounds. Most likely these sound are from the decomposition of the material as it undergoes pyrolysis (turns in gaseous fuel and char). This paper will present noise measurements from various burning articles as well as characterization of the fire to understand this noise source.

3:30

2pNSa8. Bacterial attachment and insertion loss of earplugs used longtime in the noisy workplace. Jinro Inoue, Aya Nakamura, Yumi Tanizawa, and Seichi Horie (Dept. of Health Policy and Management, Univ. of Occupational and Environ. Health, Japan, 1-1 Iseigaoka, Yahatanishi-ku, Kitakyushu, Fukuoka 807-8555, Japan, j-inoue@med.uoeh-u.ac.jp)

In the real noisy workplace, workers often use earplugs for a longtime. We assessed the condition of bacterial attachment and the insertion loss of 197 pairs of earplugs collected from 6 companies. The total viable counts and the presence of *Staphylococcus aureus* were examined by 3M Petrifilm. The insertion losses were evaluated by GRAS 45CB Acoustic Test Fixture. We detected greater number of viable counts in the foam earplugs than in the premolded earplugs. Staphylococcus aureus was detected in 10 foam earplugs (5.1%). The deterioration of insertion loss was found only in the deformed earplugs. The condition of work environment such as presence of dust or use of oily liquid might cause the deterioration. Both the condition of bacterial attachment and the insertion loss were not correlated with the duration of use. We observed no correlation between the condition of bacterial attachment and the insertion loss of earplugs and neither of them was related to the duration of long-term use of the earplugs.

TUESDAY AFTERNOON, 28 OCTOBER 2014 MARRIOTT 9/10, 1:00 P.M. TO 4:20 P.M.

Session 2pNSb

Noise and Structural Acoustics and Vibration: Launch Vehicle Acoustics II

R. Jeremy Kenny, Cochair
Marshall Flight Center, NASA, Huntsville, AL 35812

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

Chair's Introduction—1:00

Invited Papers

1:05

2pNSb1. Comparison of the acoustical emissions of multiple full-scale rocket motors. Michael M. James, Alexandria R. Salton (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, michael.james@blueridgeresearch.com), Kent L. Gee, and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Development of the next-generation space flight vehicles has prompted a renewed focus on rocket sound source characterization and near-field propagation modeling. Improved measurements of the sound near the rocket plume are critical for direct determination of the acoustical environment both in the near and far-fields. They are also crucial inputs to empirical models and to validate computational aeroacoustics models. Preliminary results from multiple measurements of static horizontal firings of Alliant Techsystems motors including the GEM-60, Orion 50S XLG, and the Reusable Solid Rocket Motor (RSRM) performed in Promontory, UT, are analyzed and compared. The usefulness of scaling by physical parameters such as nozzle diameter, velocity, and overall sound power is demonstrated. The sound power spectra, directional characteristics, distribution along the exhaust flow, and pressure statistical metrics are examined over the multiple motors. These data sets play an important role in formulating more realistic sound source models, improving acoustic load estimations, and aiding in the development of the next generation space flight vehicles via improved measurements of sound near the rocket plume.

1:25

2pNSb2. Low-dimensional acoustic structures in the near-field of clustered rocket nozzles. Andres Canchero, Charles E. Tinney (Aerosp. Eng. and Eng. Mech., The Univ. of Texas at Austin, 210 East 24th St., WRW-307, 1 University Station, C0600, Austin, TX 78712-0235, andres.canchero@utexas.edu), Nathan E. Murray (National Ctr. for Physical Acoust., Univ. of MS, Oxford, MS), and Joseph H. Ruf (NASA Marshall Space Flight Ctr., Huntsville, AL)

The plume and acoustic field produced by a cluster of two and four rocket nozzles is visualized by way of retroreflective shadowgraphy. Both steady state and transient operations of the nozzles (start-up and shut-down) were conducted in the fully-anechoic chamber and open jet facility of The University of Texas at Austin. The laboratory scale rocket nozzles comprise thrust-optimized parabolic (TOP) contours, which during start-up, experience free shock separated flow, restricted shock separated flow, and an "end-effects regime" prior to flowing full. Shadowgraphy images are first compared with several RANS simulations during steady operations. A proper orthogonal decomposition (POD) of various regions in the shadowgraphy images is then performed to elucidate the prominent features residing in the supersonic annular flow region, the acoustic near field and the interaction zone that resides between the nozzle

plumes. Synchronized surveys of the acoustic loads produced in close vicinity to the rocket clusters are compared to the low-order shadowgraphy images in order to identify the various mechanisms within the near-field that are responsible for generating sound.

1:45

2pNSb3. Experimental study on lift-off acoustic environments of launch vehicles by scaled cold jet. Hiroki Ashida, Yousuke Takeyama (Integrated Defence & Space Systems, Mitsubishi Heavy Industries, Ltd., 10, Oye-cho, Minato-ku, Nagoya City, Aichi 455-8515, Japan, hiroki1_ashida@mhi.co.jp), Kiyotaka Fujita, and Aki Azusawa (Technol. & Innovation Headquarters, Mitsubishi Heavy Industries, Ltd., Aichi, Japan)

Mitsubishi Heavy Industries (MHI) have been operating the current Japanese flagship launch vehicle H-IIA and H-IIB, and developing the next flagship launch vehicle H-X. The concept of H-X is affordable, convenient, and comfortable for payloads including mitigation of acoustic environment during launch. Acoustic measurements were conducted using scaled GN2 cold jet and aperture plate to facilitate understanding of lift-off acoustic source and to take appropriate measures against it without use of water injection. It was seen that the level of vehicle acoustics in high frequency range depends on the amount of interference between the jet and the plate, and enlargement of the aperture is effective for acoustic mitigation.

2:05

2pNSb4. Detached-Eddy simulations of rocket plume noise at lift-off. A. Lyrintzis, V. Golubev (Aerosp. Eng., Embry-Riddle Aeronutical Univ., Daytona Beach, FL), K. Kurbatski (Aerosp. Eng., Embry-Riddle Aeronutical Univ., Lebanon, New Hampshire), E. Osman (Aerosp. Eng., Embry-Riddle Aeronutical Univ., Denver, Colorado), and Reda Mankbadi (Aerosp. Eng., Embry-Riddle Aeronutical Univ., 600 S. Clyde Morris Blvd, Daytona Beach, FL 32129, Reda.Mankbadi@erau.edu)

The three-dimensional turbulent flow and acoustic field of a supersonic jet impinging on a solid plate at different inclination angles is studied computationally using the general-purpose CFD code ANSYS Fluent. A pressure-based coupled solver formulation with the second-order weighted central-upwind spatial discretization is applied. Hot jet thermal condition is considered. Acoustic radiation of impingement tones is simulated using a transient time-domain formulation. The effects of turbulence in steady state are modeled by the SST k- turbulence model. The Wall-Modeled Large-Eddy Simulation (WMLES) model is applied to compute transient solutions. The near-wall mesh on the impingement plate is fine enough to resolve the viscosity-affected near-wall region all the way to the laminar sublayer. Inclination angle of the impingement plate is parameterized in the model for automatic re-generation of the mesh and results. The transient solution reproduces the mechanism of impingement tone generation by the interaction of large-scale vortical structures with the impingement plate. The acoustic near field is directly resolved by the Computational Aeroacoustics (CAA) to accurately propagate impingement tone waves to near-field microphone locations. Results show the effect of the inclination angle on sound level pressure spectra and overall sound pressure level directivities.

2:25

2pNSb5. Large-eddy simulations of impinging over-expanded supersonic jet noise for launcher applications. Julien Troyes, François Vuillot (DSNA, Onera, BP72, 29 Ave. de la Div. Leclerc, Châtillon Cedex 92322, France, julien.troyes@onera.fr), and Hadrien Lambaré (DLA, CNES, Paris, France)

During the lift-off phase of a space launcher, powerful rocket motors generate harsh acoustic environment on the launch pad. Following the blast waves created at ignition, jet noise is a major contributor to the acoustic loads received by the launcher and its payload. Recent simulations performed at ONERA to compute the noise emitted by solid rocket motors at lift-off conditions are described. Far-field noise prediction is achieved by associating a LES solution of the jet flow with an acoustics surface integral method. The computations are carried out with in-house codes CEDRE for the LES solution and KIM for Ffowcs Williams & Hawkings porous surface integration method. The test case is that of a gas generator, fired vertically onto a 45 degrees inclined flat plate which impingement point is located 10 diameters from nozzle exit. Computations are run for varied numerical conditions, such as turbulence modeling along the plate and different porous surfaces location and type. Results are discussed and compared with experimental acoustic measurements obtained by CNES at MARTEL facility.

2:45–3:05 Break

3:05

2pNSb6. Scaling metrics for predicting rocket noise. Gregory Mack, Charles E. Tinney (Ctr. for AeroMech. Res., The Univ. of Texas at Austin, ASE/EM, 210 East 24th St., Austin, TX 78712, cetinney@utexas.edu), and Joseph Ruf (Combustion and Flow Anal. Team, ER42, NASA Marshal Space Flight Ctr., Huntsville, AL)

Several years of research at The University of Texas at Austin concerning the sound field produced by large area-ratio rocket nozzles is presented [Baars *et al*., AIAA J. 50(1), (2012); Baars and Tinney, Exp. Fluids, 54 (1468), (2013); Donald *et al*., AIAA J. 52(7), (2013)]. The focus of these studies is on developing an in-depth understanding of the various acoustic mechanisms that form during start-up of rocket engines and how they may be rendered less efficient in the generation of sound. The test articles comprise geometrically scaled replicas of large area ratio nozzles and are tested in a fully anechoic chamber under various operating conditions. A framework for scaling laboratory-scale nozzles is presented by combining established methods with new methodologies [Mayes, NASA TN D-21 (1959); Gust, NASA TN-D-1999 (1964); Eldred, NASA SP-8072 (1972); Sutherland AIAA Paper 1993–4383 (1993); Varnier, AIAA J. 39:10 (2001); James *et al*. Proc. Acoust. Soc. Amer. 18(3aNS), (2012)]. In particular, both hot and cold flow tests are reported which comprise single, three and four nozzle clusters. An effort to correct for geometric scaling is also presented.

3:25

2pNSb7. Acoustic signature characterization of a sub-scale rocket launch. David Alvord and K. Ahuja (Aerosp. & Acoust. Technologies Div., Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, david.alvord@gtri.gatech.edu)

Georgia Tech Research Institute (GTRI) conducted a flight test of a sub-scale rocket in 2013 outside Talladega, Alabama, to acquire the launch acoustics produced. The primary objective of the test was to characterize the acquired data during a sub-scale launch and compare it with heritage launch data from the STS-1 Space Shuttle flight. Neither launch included acoustic suppression; however, there were differences in the ground geometry. STS-1 launched from the Mobile Launch Platform at Pad 39B with the RS-25 liquid engines and Solid Rocket Boosters (SRBs) firing into their respective exhaust ducts and flame trench, while the GTRI flight test vehicle launched from a flat reflective surface. The GTRI launch vehicle used a properly scaled Solid Rocket Motor (SRM) for propellant; therefore, primary analysis will focus on SRM/SRB centric acoustic events. Differences in the Ignition Overpressure (IOP) wave signature between both due to this will be addressed. Additionally, the classic liftoff acoustics "football shape" is preserved between both full and sub-scale flights. The launch signatures will be compared, with note taken of specific launch acoustic events more easily investigated with sub-scale launch data or supplement current sub-scale static hotfire testing.

3:45

2pNSb8. Infrasonic energy from orbital launch vehicles. W. C. Kirkpatrick Alberts, John M. Noble, and Stephen M. Tenney (US Army Res.Lab., 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net)

Large, heavy-lift rockets have significant acoustic and infrasonic energy that can often be detected from a considerable distance. These sounds, under certain environmental conditions, can propagate hundreds of kilometers from the launch location. Thus, ground-based infrasound arrays can be used to monitor the low frequencies emitted by these large rocket launches. Multiple launches and static engine tests have been successfully recorded over many years using small infrasound arrays at various distances from the launch location. Infrasonic measurements using a 20 m array and parabolic equation modeling of a recent launch of an Aries V rocket at Wallops Island, Virginia, will be discussed.

Contributed Paper

4:05

2pNSb9. Influence of source level, peak frequency, and atmospheric absorption on nonlinear propagation of rocket noise. Michael F. Pearson (Phys., Brigham Young Univ., 560 W 700 S, Lehi, UT 84043, m3po22@gmail.com), Kent L. Gee, Tracianne B. Neilsen, Brent O. Reichman (Phys., Brigham Young Univ., Provo, UT), Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC), and Alexandira R. Salton (Blue Ridge Res. and Consulting, Asheville, Utah)

Nonlinear propagation effects in rocket noise have been previously shown to be significant [M. B. Muhlestein *et al*. Proc. Mtgs. Acoust. (2013)]. This paper explores the influence of source level, peak frequency, and ambient atmospheric conditions on predictions of nonlinear propagation. An acoustic pressure waveform measured during a full-scale solid rocket motor firing is numerically propagated via generalized Burgers equation model for atmospheric conditions representative of plausible spaceport locations. Cases are explored where the overall sound pressure level and peak frequency has been scaled to model engines of different scale or thrust. The predicted power spectral densities and overall sound pressure levels, both flat and A-weighted, are compared for nonlinear and linear propagation for distances up to 30 km. The differences in overall level suggest that further research to appropriately include nonlinear effects in launch vehicle noise models is worthwhile.

Session 2pPA

Physical Acoustics and Education in Acoustics: Demonstrations in Acoustics

Uwe J. Hansen, Cochair
Chemistry & Physics, Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

Murray S. Korman, Cochair
Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402

Chair's Introduction—1:00

Invited Papers

1:05

2pPA1. Sharing experiments for home and classroom demonstrations. Thomas D. Rossing (Stanford Univ., Music, Stanford, CA 94305, rossing@ccrma.stanford.edu)

In the third edition of The Science of Sound, we included a list of "Experiments for Home, Laboratory, and Classroom Demonstrations" at the end of each chapter. Some of the demonstrations are done by the instructor in class, some are done by students for extra credit, some are intended to be done at home. We describe a representative number of these, many of which can be done without special equipment.

1:30

2pPA2. A qualitative demonstration of the behavior of the human cochlea. Andrew C. Morrison (Dept. of Natural Sci., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

Demonstrations of the motion of the basilar membrane in the human cochlea designed by Keolian [J. Acoust. Soc. Am. **101**, 1199-1201 (1997)], Tomlinson et. al. [J. Acoust. Soc. Am. **121**, 3115 (2007)], and others provide a way for students in a class to visualize the behavior of the basilar membrane and explore the physical mechanisms leading to many auditory phenomena. The designs of Keolian and Tomlinson are hydrodynamic. A non-hydrodynamic apparatus has been designed that can be constructed with commonly available laboratory supplies and items readily available at local hardware stores. The apparatus is easily set up for demonstration purposes and is compact for storing between uses. The merits and limitations of this design will be presented.

1:55

2pPA3. Nonlinear demonstrations in acoustics. Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

The world is nonlinear, and in presenting demonstrations in acoustics, one often has to consider the effects of nonlinearity. In this presentation the nonlinear effects are made to be very pronounced. The nonlinear effects of standing waves on a lightly stretched string (which is also very elastic) lead to wave shape distortion, mode jumping and hysteresis effects in the resonant behavior of a tuning curve near a resonance. The effects of hyperelasticity in a rubber string are discussed. A two dimensional system like a vibrating rectangular or circular drum-head are well known. The nonlinear effects of standing waves on a lightly stretched hyperelastic membrane make an interesting and challenging study. Here, tuning curve behavior demonstrates that there is softening of the system for slightly increasing vibration amplitudes followed by stiffening of the system at larger vibration amplitudes. The hysteretic behavior of the tuning curve for sweeping from lower to higher frequencies and then from higher to lower frequencies (for the same drive amplitude) is demonstrated. Lastly, the nonlinear effects of a column of soil or fine granular material loading a thin elastic circular clamped plate are demonstrated near resonance. Here again, the nonlinear highly asymmetric tuning curve behavior is demonstrated.

2:20–2:40 Audience Interaction

Session 2pSA

Structural Acoustics and Vibration, Signal Processing in Acoustics, and Engineering Acoustics: Nearfield Acoustical Holography

Sean F. Wu, Chair
Mechanical Engineering, Wayne State University, 5050 Anthony Wayne Drive, College of Engineering Building, Rm 2133, Detroit, MI 48202

Chair's Introduction—2:00

Invited Papers

2:05

2pSA1. Transient nearfield acoustical holography. Sean F. Wu (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., College of Eng. Bldg., Rm. 2133, Detroit, MI 48202, sean_wu@wayne.edu)

Transient Nearfield Acoustical Holography Sean F. Wu Department of Mechanical Engineering, Wayne State University, Detroit, MI, 48202 This paper presents the general formulations for reconstructing the transient acoustic field generated by an arbitrary object with a uniformly distributed surface velocity in free space. These formulations are derived from the Kirchhoff-Helmholtz integral theory that correlates the transient acoustic pressure at any field point to those on the source surface. For a class of acoustic radiation problems involving an arbitrarily oscillating object with a uniformly distributed surface velocity, for example, a loudspeaker membrane, the normal surface velocity is frequency dependent but is spatially invariant. Accordingly, the surface acoustic pressure is expressible as the product of the surface velocity and the quantity that can be solved explicitly by using the Kirchhoff-Helmholtz integral equation. This surface acoustic pressure can be correlated to the field acoustic pressure using the Kirchhoff-Helmholtz integral formulation. Consequently, it is possible to use nearfield acoustic holography to reconstruct acoustic quantities in entire three-dimensional space based on a single set of acoustic pressure measurements taken in the near field of the target object. Examples of applying these formulations to reconstructing the transient acoustic pressure fields produced by various arbitrary objects are demonstrated.

2:30

2pSA2. A multisource-type representation statistically optimized near-field acoustical holography method. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

A reduced-order approach to near-field acoustical holography (NAH) that accounts for sound fields generated by multiple spatially separated sources of different types is presented. In this method, an equivalent wave model (EWM) of a given field is formulated based on rudimentary knowledge of source types and locations. The statistically optimized near-field acoustical holography (SONAH) algorithm is utilized to perform the NAH projection after the formulation of the multisource EWM. The combined process is called multisource-type representation SONAH (MSTR SONAH). This method is used to reconstruct simulated sound fields generated by combinations of multiple source types. It is shown that MSTR SONAH can successfully reconstruct the near field pressures in multisource environments where other NAH methods result in large errors. The MSTR SONAH technique can be extended to general sound fields where the shapes and locations of sources and scattering bodies are known.

2:55

2pSA3. Bayesian regularization applied to real-time near-field acoustic holography. Thibaut Le Magueresse (MicrodB, 28 chemin du petit bois, Ecully 69131, France, thibaut-le-magueresse@microdb.fr), Jean-Hugh Thomas (Laboratoire d'Acoustique de l'Université du Maine, Le Mans, France), Jérôme Antoni (Laboratoire Vibrations Acoustique, Villeurbanne, France), and Sébasien Paillasseur (MicrodB, Ecully, France)

Real-Time Near-field Acoustic Holography is used to recover non stationary acoustic sound sources using a planar microphone array. In the direct way, describing propagation requires the convolution of the spatial spectrum of the source under study with a known impulse response. When the convolution operator is replaced with a matrix product, the propagation operator is re-written in a Toeplitz matrix form. Solving the inverse problem is based on a Singular value decomposition of this propagator and Tikhonov regularization is used to stabilize the solution. The purpose here is to study the regularization process. The formulation of this problem in the Tikhonov sense estimates the solution from the knowledge of the propagation model, the measurements and the regularization parameter. This parameter is calculated by making a compromise between the fidelity to the real measured data and the fidelity to available a priori information. A new regularization parameter is introduced based on a Bayesian approach to maximize the information taken into account. Comparisons of the results are proposed, using the L-Curve and the generalized cross validation. The superiority of the Bayesian parameter is observed for the reconstruction of a non stationary experimental source using real-time near-field acoustic holography.

Contributed Papers

3:20

2pSA4. Acoustic building infiltration measurement system. Ralph T. Muehleisen, Eric Tatara (Decision and Information Sci., Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov), Ganesh Raman, and Kanthasamy Chelliah (Mater., Mech., and Aerosp. Eng., Illinois Inst. of Technol., Chicago, IL)

Building infiltration is a significant portion of the heating and cooling load of buildings and accounts for nearly 4% of the total energy use in the United States. Current measurement methods for locating and quantifying infiltration in commercial buildings to apply remediation are very limited. In this talk, the development of a new measurement system, the Acoustic Building Infiltration Measurement System (ABIMS) is presented. ABIMS uses Nearfield Acoustic Holography (NAH) to measure the sound field transmitted through a section of the building envelope. These data are used to locate and quantify the infiltration sites of a building envelope section. The basic theory of ABIMS operation and results from computer simulations are presented.

3:35

2pSA5. Reversible quasi-holographic line-scan processing for acoustic imaging and feature isolation of transient scattering. Daniel Plotnick, Philip L. Marston, David J. Zartman (Phys. and Astronomy, Washington State Univ., 1510 NW Turner DR, Apartment 4, Pullman, WA 99163, dsplotnick@gmail.com), and Timothy M. Marston (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Transient acoustic scattering data from objects obtained using a one-dimensional line scan or two-dimensional raster scan can be processed via a linear quasi-holographic method [K. Baik, C. Dudley, and P. L. Marston, J. Acoust. Soc. Am. 130, 3838–3851 (2011)] in a way that is reversible, allowing isolation of spatially or temporally dependent features [T. M. Marston *et al.*, in Proc. IEEE Oceans 2010]. Unlike nearfield holography the subsonic wavenumber components are suppressed in the processing. Backscattering data collected from a collocated source/receiver (monostatic scattering) and scattering involving a stationary source and mobile receiver (bistatic) may be processed in this manner. Distinct image features such as those due to edge diffraction, specular reflection, and elastic effects may be extracted in the image domain and then reverse processed to allow examination of those features in time and spectral domains. Multiple objects may also be isolated in this manner and clutter may be removed [D. J. Zartman, D. S. Plotnick, T. M. Marston, and P. L. Marston, Proceedings of Meetings on Acoustics 19, 055011 (2013) http://dx.doi.org/10.1121/1.4800881]. Experimental examples comparing extracted features with physical models will be discussed and demonstrations of signal enhancement in an at sea experiment, TREX13, will be shown. [Work supported by ONR.]

TUESDAY AFTERNOON, 28 OCTOBER 2014 — MARRIOTT 5, 1:00 P.M. TO 5:00 P.M.

Session 2pSC

Speech Communication: Segments and Suprasegmentals (Poster Session)

Olga Dmitrieva, Chair

Purdue University, 640 Oval Drive, Stanley Coulter 166, West Lafayette, IN 47907

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

Contributed Papers

2pSC1. Interactions among lexical and discourse characteristics in vowel production. Rachel S. Burdin, Rory Turnbull, and Cynthia G. Clopper (Linguist, The Ohio State Univ., 1712 Neil Ave., 222 Oxely Hall, Columbus, OH 43210, burdin@ling.osu.edu)

Various factors are known to affect vowel production, including word frequency, neighborhood density, contextual predictability, mention in the discourse, and audience. This study explores interactions between all five of these factors on vowel duration and dispersion. Participants read paragraphs that contained target words which varied in predictability, frequency, and density. Each target word appeared twice in the paragraph. Participants read each paragraph twice: as if they were talking to a friend ("plain speech") and as if they were talking to a hearing-impaired or non-native interlocutor ("clear speech"). Measures of vowel duration and dispersion were obtained. Results from the plain speech passages revealed that second mention and more predictable words were shorter than first mention and less predictable words, and that vowels in second mention and low density words were less peripheral than in first mention and high density words. Interactions between frequency and mention, and density and mention, were also observed, with second mention reduction only occurring in low density and low frequency words. We expect to observe additional effects of speech style, with clear speech vowels being longer and more disperse than plain speech vowels, and that these effects will interact with frequency, density, predictability, and mention.

2pSC2. Phonetic correlates of phonological quantity of Yakut. Lena Vasilyeva, Juhani Järvikivi, and Anja Arnhold (Dept. of Linguist, Univ. of AB, Edmonton, AB T6G2E7, Canada, lvasilye@ualberta.ca)

We investigated vowel quantity in Yakut (Sakha), a Turkic language spoken in Siberia by over 400,000 speakers in the Republic of Sakha (Yakutia) in the Russian Federation. Yakut is a quantity language; all vowel and consonant phonemes have short and long contrastive counterparts. The study aims at revealing acoustic characteristics of the binary quantity distinction in vowels. We used two sets of data: (1) A female native Yakut speaker read a 200-word list containing disyllabic nouns and verbs with four different combinations of vowel length in the two syllables (short–short, short–long, long–short, and long–long) and a list of 50 minimal pairs differing only in vowel length; (2) Spontaneous speech data from 9 female native Yakut speakers (aged 19–77), 200 words with short vowels and 200 words with long vowels, were extracted for analysis. Acoustic measurements of the short and long vowels' f0-values, duration and intensity were done. Mixed-effects models showed a significant durational difference between long and short vowels for both data sets. However, the preliminary results indicated that, unlike in quantity languages like Finnish and Estonian, there was no consistent effect of f0 as the phonetic correlate in Yakut vowel quantity distinction.

2pSC3. Acoustic and perceptual characteristics of vowels produced by self-identified gay and heterosexual male speakers. Keith Johnson (Linguist, Univ. of California, Berkeley, Berkeley, CA) and Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research (Tracy & Satariano, 2011) investigated the perceptual characteristics of gay and heterosexual male speech; it was discovered that listeners primarily relied on vowels to identify sexual orientation. Using single-word utterances produced by those same speakers, the current study examined both the acoustic characteristics of vowels, such as pitch, duration, and the size of the vowel space, and how these characteristics relate to the perceived sexual orientation of the speaker. We found a correlation between pitch and perceived sexual identity for vowels produced by heterosexual speakers—higher f0 was associated with perceptual "gayness." We did not find this correlation for gay speakers. Vowel duration did not reliably distinguish gay and heterosexual speakers, but speakers who produced longer vowels were perceived as gay and speakers who produced shorter vowels were perceived as heterosexual. The size of the vowel space did not reliably differ between gay and heterosexual speakers. However, speakers who produced a larger vowel space were perceived as more gay-sounding than speakers who produced a smaller vowel space. The results suggest that listeners rely on these acoustic characteristics when asked to determine a male speaker's sexual orientation, but that the stereotypes that they seem to rely upon are inaccurate.

2pSC4. Acoustic properties of the vowel systems of Bolivian Quechua/Spanish bilinguals. Nicole Holliday (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, nrh245@nyu.edu)

This paper describes the vowel systems of Quechua/Spanish bilinguals in Cochabamba, Bolivia, and examines these systems to illuminate variation between phonemic and allophonic vowels in this Quechua variety. South Bolivian Quechua is described as phonemically trivocalic, and Bolivian Spanish is described as pentavocalic (Cerrón-Palomino 1994). Although South Bolivian Quechua has three vowel categories, Quechua uvular stop consonants promote high vowel lowering, effectively lowering /i/ and /u/ towards space otherwise occupied by /e/ and /o/ respectively, producing a system with five surface vowels but three phonemic vowels (Buckley 2000). The project was conducted with eleven Quechua/Spanish bilinguals from the Cochabamba department in Bolivia. Subjects participated in a Spanish to Quechua oral translation task and a word list task in Spanish. Results indicate that Quechua/Spanish bilinguals maintain separate vowel systems. In the Spanish vowel systems, each vowel occupies its own space and backness. A one-way ANOVA reveals that /i/ is higher and fronter than /e/, and /u/ is higher than /o/ ($p<0.05$). The Quechua vowel systems are somewhat more variable, with substantial overlap between /i/ and /e/, and between /u/ and /o/. Potential explanations for this result include lexical conditioning, speaker literacy effects, and differences in realizations of phonemic versus allophonic vowels.

2pSC5. Cue integration in the perception of fricative-vowel coarticulation in Korean. Goun Lee and Allard Jongman (Linguist, The Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045-3129, cconni@ku.edu)

Korean distinguishes two fricatives—fortis [s'] and non-fortis [s]. Perception of this distinction was tested in two different vowel contexts, with three types of stimuli (consonant-only, vowel-only, or consonant-vowel sequences) (Experiment 1). The relative contribution of consonantal and vocalic cues was also examined with cross-spliced stimuli (Experiment 2). Listeners' weighting of 7 perceptual cues—spectral mean (initial 60%, final 40%), vowel duration, H1-H2* (onset, mid), and cepstral peak prominence (onset, mid)—was examined. The data demonstrate that identification performance was heavily influenced by vowel context and listener performance was more accurate in the /a/ vowel context than in the /i/ vowel context. In addition, the type of stimulus presented changed the perceptual cue weighting. When presented with conflicting cues, listener decisions were driven by the vocalic cues in the /a/ vowel context. These results suggest that perceptual cues associated with breathy phonation are the primary cues for fricative identification in Korean.

2pSC6. Voicing, devoicing, and noise measures in Shanghainese voiced and voiceless glottal fricatives. Laura L. Koenig (Haskins Labs and Long Island Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu) and Lu-Feng Shi (Haskins Labs and Long Island Univ., Brooklyn, New York)

Shanghainese has a rather rare voicing distinction between the glottal fricatives /h/ and /ɦ/. We evaluate the acoustic characteristics of this contrast in ten male and ten female speakers of urban Shanghainese dialect. Participants produced 20 CV words with a mid/low central vowel in a short carrier phrase. All legal consonant-tone combinations were used: /h/ preceded high, low, and short tones whereas /ɦ/ preceded low and short tones. Preliminary analyses suggested that the traditional "voiced" and "voiceless" labels for these sounds are not always phonetically accurate; hence we measure the duration of any voicing break relative to the entire phrase, as well as the harmonics-to-noise ratio (HNR) over the time. We expect longer relative voiceless durations and lower HNR measures for /h/ compared to /ɦ/. A question of interest is whether any gender differences emerge. A previous study on American English [Koenig, 2000, JSLHR 43, 1211–1228] found that men phonated through their productions of /h/ more often than women, and interpreted that finding in terms of male-female differences in vocal fold characteristics. A language that contrasts /h/ and /ɦ/ might minimize any such gender variation. Alternatively, the contrast might be realized in slightly different ways in men and women.

2pSC7. Incomplete neutralization of sibilant consonants in Penang Mandarin: A palatographic case study. Ting Huang, Yueh-chin Chang, and Feng-fan Hsieh (Graduate Inst. of Linguist, National Tsing Hua Univ., Rm. B306, HSS Bldg., No. 101, Section 2, Kuang-Fu Rd., Hsinchu City 30013, Taiwan, funting.huang@gmail.com)

It has been anecdotally observed that the three-way contrasts in Standard Chinese are reduced to two-way contrasts in Penang Mandarin (PM). PM is a variety of Mandarin Chinese spoken in Penang of Malaysia, which is influenced by Penang Hokkien. This work shows that the alleged neutralization of contrasts is incomplete (10 consonants x 3 vowel contexts x 5 speakers). More specifically, alveopalatal [ɕ] may range from postalveolar zone (73.33%) to alveolar zone (26.67%), and so does retroflex [ʂ] (46.67% vs. 46.67%). [s] and [n] are apical (or [+anterior]) coronals. The goal of this study is three-fold: (i) to describe the places of articulation of PM coronals and the patterns of ongoing sound changes, (ii) to show the neutralization of place contrasts is incomplete whereby constriction length remains distinct for these sibilant sounds, and (iii) to demonstrate different coarticulatory patterns of consonants in variant vowel contexts. The intricate division of coronal consonants does not warrant a precise constriction location on the upper palate. This PM data lend support to Ladefoged and Wu's (1984) observation that it is not easy to pin down a clear-cut boundary between dental and alveolar stops, and between alveolar and palatoalveolar fricatives.

2pSC8. Final voicing and devoicing in American English. Olga Dmitrieva (Linguistics/School of Lang. and Cultures, Purdue Univ., 100 North University St., Beering Hall, Rm. 1289, West Lafayette, IN 47907, odmitrie@purdue.edu)

English is typically described as a language in which voicing contrast is not neutralized in word-final position. However, a tendency towards devoicing (at least partial) of final voiced obstruents in English has been reported by the previous studies (e.g., Docherty (1992) and references therein). In the present study, we examine a number of acoustic correlates of obstruent voicing and the robustness with which each one is able to differentiate between voiced and voiceless obstruents in the word-final position in the speech recorded by twenty native speakers of the Mid-Western dialect of American English. The examined acoustic properties include preceding vowel duration, closure or frication duration, duration of the release portion, and duration of voicing during the obstruent closure, frication, and release. Initial results indicate that final voiced obstruents are significantly different from the voiceless ones in terms of preceding vowel duration and closure/frication duration. However, release duration for stops does not appear to correlate with voicing in an equally reliable fashion. A well-pronounced difference in terms of closure voicing between voiced and voiceless final stops is significantly reduced in fricative consonants, which indicates a tendency towards neutralization of this particular correlate of voicing in the word-final fricatives of American English.

2pSC9. An analysis of the singleton-geminate contrast in Japanese fricatives and stops. Christopher S. Rourke and Zack Jones (Linguist, The Ohio State Univ., 187 Clinton St., Columbus, OH 43202, rourke.16@osu.edu)

Previous acoustic analyses of the singleton-geminate contrast in Japanese have focused primarily on read speech. The present study instead analyzed the lengths of singleton and geminate productions of word-medial fricatives and voiceless stops in spontaneous monologues from the Corpus of Spontaneous Japanese (Maekawa, 2003). The results of a linear mixed effects regression model mirrored previous findings in read speech that the geminate effect (the durational difference between geminate and singletons) of stops is significantly larger than that of fricatives. This study also found a large range of variability in the geminate effect size between talkers. The size of the geminate effect between fricatives and voiceless stops was found to be slightly correlated, suggesting that they might be related to other rate-associated production differences between individuals. This suggestion was evaluated by exploring duration differences associated with talker age and gender. While there was no relationship between age and duration, males produced shorter durations than females for both fricatives and stops. However, the size of the geminate effect was not related to the gender of the speaker. The cause of these individual differences may be related to sound perception. Future research will investigate the cause of these individual differences in geminate effect size.

2pSC10. Quantifying surface phonetic variation using acoustic landmarks as feature cues. Jeung-Yoon Choi and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, MIT, 50 Vassar St., Rm. 36-523, Cambridge, MA 02139, sshuf@mit.edu)

Acoustic landmarks, which are abrupt spectral changes associated with certain feature sequences in spoken utterances, are highly informative and have been proposed as the initial analysis stage in human speech perception, and for automatic speech recognition (Stevens, JASA 111(4), 2002, 1872–1891). These feature cues and their parameter values also provide an effective tool for quantifying systematic context-governed surface phonetic variation (Shattuck-Hufnagel and Veilleux, ICPhS XVI, 2007, 925–928). However, few studies have provided landmark-based information about the full range of variation in continuous communicative speech. The current study examines landmark modification patterns in a corpus of maptask-elicited speech, hand annotated for whether the landmarks were realized as predicted from the word forms or modified in context. Preliminary analyses of a single conversation (400 s, one speaker) show that the majority of landmarks (about 84%) exhibited the canonical form predicted from their lexical specifications, and that modifications were distributed systematically across segment types. For example, 90% of vowel landmarks (at amplitude/F1 peaks) were realized as predicted, but only 70% of the closures for non-strident fricatives /v/ and /dh/, and 40–50% of /t/ closures and releases. Further quantification of landmark modification patterns will provide useful information about the processing of surface phonetic variation.

2pSC11. Age- and gender-related variation in voiced stop prenasalization in Japanese. Mieko Takada (Aichi Gakuin Univ., Nisshin, Japan), Eun Jong Kong (Korea Aerosp. Univ., Goyang-City, South Korea), Kiyoko Yoneyama (Daito Bunka Univ., 1-9-1 Takashimadaira, Itabashi-ku, Tokyo 175-8571, Japan, yoneyama@ic.daito.ac.jp), and Mary E. Beckman (Ohio State Univ., Columbus, OH)

Modern Japanese is generally described as having phonologically voiced (versus voiceless) word-initial stops. However, phonetic details vary across dialects and age groups; in Takada's (2011) measurements of recordings of 456 talkers from multiple generations of talkers across five dialects, Osaka-area speakers and older speakers in the Tokyo area (Tokyo, Chiba, Saitama, and Kanagawa prefectures) typically show pre-voicing (lead VOT), but younger speakers show many "devoiced" (short lag VOT) values, a tendency that is especially pronounced among younger Tokyo-area females. There is also variation in the duration of the voice bar, with very long values (up to -200 ms lead VOT) observed in the oldest female speakers. Spectrograms of such tokens show faint formants during the stop closure, suggesting a velum-lowering gesture to vent supra-glottal air pressure to sustain vocal fold vibration. Further evidence of pre-nasalization in older Tokyo-area females comes from comparing amplitude trajectories for the voice bar to amplitude trajectories during nasal consonants, adapting a method proposed by Burton, Blumstein, and Stevens (1972) for exploring phonemic pre-nasalization contrasts. Differences in trajectory shape patterns between the oldest males and females and between older and younger females are like the differences that Kong, Syrika, and Edwards (2012) observed across Greek dialects.

2pSC12. An acoustic comparison of dental and retroflex sibilants in Chinese Mandarin and Taiwan Mandarin. Hanbo Yan and Allard Jongman (Linguist, Univ. of Kansas, 1732 Anna Dr., Apt. 11, Lawrence, KS 66044, yanhanbo@ku.edu)

Mandarin has both dental and retroflex sibilants. While the Mandarin varieties spoken in China and Taiwan are often considered the same, native speakers of Mandarin can tell the difference between the two. One obvious difference is that between the retroflex ([ʂ], [tʂ], [tʂh]) and dental sibilants ([s], [ts], [tsh]). This study investigates the acoustic properties of the sibilants of Chinese Mandarin and Taiwan Mandarin. Eight native speakers each of Chinese and Taiwan Mandarin produced the six target sibilants in word-initial position. A number of acoustic parameters, including spectral moments and duration, were analyzed to address two research questions: (a) which parameters distinguish the dental and retroflex in each type of Mandarin; (b) is there a difference between Chinese and Taiwan Mandarin? Results show that retroflex sibilants have a lower M1 and M2, and a higher M3 than dental sibilants in each language. Moreover, Chinese Mandarin has significantly larger M1, M2, and M3 differences than Taiwan Mandarin. This pattern suggests that, in contrast to Chinese Mandarin, Taiwan Mandarin is merging the retroflex sibilants in a dental direction.

2pSC13. Statistical relationships between phonological categories and acoustic-phonetic properties of Korean consonants. Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., 344 French East Bldg., Cincinnati, OH 45267, noah.silbert@uc.edu) and Hanyong Park (Linguist, Univ. of Wisconsin, Milwaukee, WI)

The mapping between segmental contrasts and acoustic-phonetic properties is complex and many-to-many. Contrasts are often cued by a multiple acoustic-phonetic properties, and acoustic-phonetic properties typically provide information about multiple contrasts. Following the approach of de Jong *et al.* (2011, JASA 129, 2455), we analyze multiple native speakers' repeated productions of Korean obstruents using a hierarchical multivariate statistical model of the relationship between multidimensional acoustics and phonological categories. Specifically, we model the mapping between categories and multidimensional acoustic measurements from multiple repetitions of 14 Korean obstruent consonants produced by 20 native speakers (10 male, 10 female) in onset position in monosyllables. The statistical model allows us to analyze distinct within- and between-speaker sources of variability in consonant production, and model comparisons allow us to assess the utility of complexity in the assumed underlying phonological category system. In addition, by using the same set of acoustic measurements for the current project's Korean consonants and the English consonants analyzed by de Jong *et al.*, we can model the within- and between-language acoustic similarity of phonological categories, providing a quantitative basis for predictions about cross-language phonetic perception.

2pSC14. Corpus testing a fricative discriminator: Or, just how invariant is this invariant? Philip J. Roberts (Faculty of Linguist, Univ. of Oxford, Ctr. for Linguist and Philology, Walton St., Oxford OX1 2HG, United Kingdom, philip.roberts@ling-phil.ox.ac.uk), Henning Reetz (Institut fuer Phonetik, Goethe-Universitaet Frankfurt, Frankfurt-am-Main, Germany), and Aditi Lahiri (Faculty of Linguist, Univ. of Oxford, Oxford, United Kingdom)

Acoustic cues to the distinction between sibilant fricatives are claimed to be invariant across languages. Evers *et al*. (1998) present a method for distinguishing automatically between [s] and [ʃ], using the slope of regression lines over separate frequency ranges within a DFT spectrum. They report accuracy rates in excess of 90% for fricatives extracted from recordings of minimal pairs in English, Dutch and Bengali. These findings are broadly replicated by Maniwa *et al*. (2009), using VCV tokens recorded in the lab. We tested the algorithm from Evers *et al*. (1998) against tokens of fricatives extracted from the TIMIT corpus of American English read speech, and the Kiel corpora of German. We were able to achieve similar accuracy rates to those reported in previous studies, with the following caveats: (1) the measure relies on being able to perform a DFT for frequencies from 0 to 8 kHz, so that a minimum sampling rate of 16 kHz is necessary for it to be effective, and (2) although the measure draws a similarly clear distinction between [s] and [ʃ] to those found in previous studies, the threshold value between the two sounds is sensitive to the dynamic range of the input signal.

2pSC15. Discriminant variables for plosive- and fricative-type single and geminate stops in Japanese. Shigeaki Amano and Kimiko Yamakawa (Faculty of Human Informatics, Aichi Shukutoku Univ., 2-9 Katahira, Nagakute, Aichi 480-1197, Japan, psy@asu.aasa.ac.jp)

Previous studies suggested that a plosive-type geminate stop in Japanese is discriminated from a single stop with variables of stop closure duration and subword duration that spans from the mora preceding the geminate stop to the vowel following the stop. However, this suggestion does not apply to a fricative-type geminate stop that does not have a stop closure. To overcome this problem, this study proposes Inter-Vowel Interval (IVI) and Successive Vowel Interval (SVI) as discriminant variables. IVI is the duration between the end of the vowel preceding the stop and the beginning of the vowel following the stop. SVI is the duration between the beginning of the vowel preceding the stop and the end of the vowel following the stop. When discriminant analysis was conducted between single and geminate stops of plosive and fricative types using IVI and SVI as independent variables, the discriminant ratio was very high (99.5%, n = 368). This result indicates that IVI and SVI are the general variables that represent acoustic features distinguishing Japanese single and geminate stops of both plosive and fricative types. [This study was supported by JSPS KAKENHI Grant Numbers 24652087, 25284080, 26370464 and by Aichi-Shukutoku University Cooperative Research Grant 2013-2014.]

2pSC16. Perceptual affinity of Mandarin palatals and retroflexes. Yung-hsiang Shawn Chang (Dept. of English, National Taipei Univ. of Technol., Zhongxiao E. Rd., Sec. 3, No. 1, Taipei 106, Taiwan, shawnchang@ntut.edu.tw)

Mandarin palatals [tɕ, tɕʰ, ɕ], which only occur before [i, y] vowels, are in complementary distribution with the alveolars [ts, tsʰ, s], the velars [k, kʰ, x], and the retroflexes [tʂ, tʂʰ, ʂ]. Upon investigating perceptually motivated accounts for the phonological representation of the palatals, Wan (2010) reported that Mandarin palatals were considered more similar to the alveolars than the velars, whereas Lu (2014) found categorical results for the palatal-alveolar discrimination. The current study furthered the investigation to the perceptual affinity between Mandarin palatals and retroflexes by having 15 native listeners identify two 8-step resynthesized [ʂ-s] continua (adapted from Chang *et al*. (2013)) cross-spliced with [i, y] vowels, respectively. To avoid phonotactic restrictions from biasing perception, all listeners were trained on identifying the [ɕi, ʂi, si] and [ɕy, ʂy, sy] syllables produced by a phonetician before the experiment. The results showed that all resynthesized stimuli, though lacking palatal-appropriate vocalic transitions, were subject to palatal perception. Particularly, two intermediate steps along the [ʂi-si] continuum and five along the [ʂy-sy] continuum were identified as palatal syllables by over 70% of the listeners. The results suggest that Mandarin palatals could be identified with both the retroflexes and alveolars based on perceptual affinity.

2pSC17. Perceptual distinctiveness of dental vs. palatal sibilants in different vowel contexts. Mingxing Li (Linguist, The Univ. of Kansas, 1541 Lilac Ln., Blake Hall 427, Lawrence, KS 66045-3129, mxlistar@gmail.com)

This paper reports a similarity rating experiment and a speeded AX discrimination experiment to test the perceptual distinctiveness of dental vs. palatal sibilants in different vowel contexts. The stimuli were pairs of CV sequences where the onsets were [s, ts, tsʰ] vs. [ɕ, tɕ, tɕʰ] as in Mandarin Chinese and the vowels were [i, a, o]; the durations of the consonants and vowels were set to values close to those in natural speech; the inter-stimulus-interval was set at 100ms to facilitate responses based on psychoacoustic similarity. A significant effect of vowel contexts was observed in the similarity rating by 20 native American English speakers, whereby the dental vs. palatal sibilants were judged to be the least distinct in the [i] context. A similar pattern was observed in the speeded AX discrimination, whereby the [i] context introduced slower "different" responses than other vowels. In general, this study supports the view that the perceptual distinctiveness of a consonant pair may vary with different vowel contexts. Moreover, the experiment results match the typological pattern of dental vs. palatal sibilants across Chinese dialects, where contrasts like /si, tsi, tsʰi/ vs. /ɕi, tɕi, tɕʰi/ are often enhanced by vowel allophony.

2pSC18. Phonetic correlates of stance-taking. Valerie Freeman, Richard Wright, Gina-Anne Levow (Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195, valerief@uw.edu), Yi Luan (Elec. Eng., Univ. of Washington, Seattle, WA), Julian Chan (Linguist, Univ. of Washington, Seattle, WA), Trang Tran, Victoria Zayats (Elec. Eng., Univ. of Washington, Seattle, WA), Maria Antoniak (Linguist, Univ. of Washington, Seattle, WA), and Mari Ostendorf (Elec. Eng., Univ. of Washington, Seattle, WA)

Stance, or a speaker's attitudes or opinions about the topic of discussion, has been investigated textually in conversation- and discourse analysis and in computational models, but little work has focused on its acoustic-phonetic properties. This is a difficult problem, given that stance is a complex activity that must be expressed along with several other types of meaning (informational, social, etc.) using the same acoustic channels. In this presentation, we begin to identify some acoustic indicators of stance in natural speech using a corpus of collaborative conversational tasks which have been hand-annotated for stance strength (none, weak, moderate, and strong) and polarity (positive, negative, and neutral). A preliminary analysis of 18 dyads completing two tasks suggests that increases in stance strength are correlated with increases in speech rate and pitch and intensity medians and ranges. Initial results for polarity also suggest correlations with speech rate and intensity. Current investigations center on local modulations in pitch and intensity, durational and spectral differences between stressed and unstressed vowels, and disfluency rates in different stance conditions. Consistent male/female differences are not yet apparent but will also be examined further.

2pSC19. Compounds in modern Greek. Angeliki Athanasopoulou and Irene Vogel (Linguist and Cognit. Sci., Univ. of Delaware, 125 East Main St., Newark, DE 19716, angeliki@udel.edu)

The difference between compounds and phrases has been studied extensively in English (e.g., Farnetani, Torsello, & Cosi, 1988; Plag, 2006; Štekauer, Zimmermann, & Gregová, 2007). However, little is known about the analogous difference in Modern Greek (Tzakosta, 2009). Greek compounds (Ralli, 2003) form a single phonological word, and thus, they only contain one primary stress. That means that the individual words lose their primary stress. The present study is the first acoustic investigation of the stress properties of Greek compounds and phrases. Native speakers of Greek produce ten novel adjective + noun compounds and their corresponding phrases (e.g., phrase: [kócino δódi] "a red tooth" vs. compound: [kocinoδódis] "someone with red teeth") in the sentence corresponding to "The XXX is at the top/bottom of the screen." Preliminary results confirm the earlier descriptive claims that compounds only have a single stress, while phrases have one on each word. Specifically, the first word (i.e., adjective) in compounds is reduced in F0 (101 Hz), duration (55 ms), and intensity (64 dB) compared to phrases (F0 = 117Hz, duration = 85 ms, and intensity = 67 dB). Also, both words are very similar for all of the measures in phrases. The second word (i.e., noun) is longer than the first word, possibly indicating phrase-final lengthening.

2pSC20. Evoked potentials during voice error detection at register boundaries. Anjli Lodhavia (Dept. of Commun. Disord. and Sci., Rush Univ., 807 Reef Court, Wheeling, IL 60090, alodhavia1@gmail.com), Sona Patel (Dept. of Speech-Lang. Pathol., Seton Hall Univ., South Orange, NJ), Saul Frankford (Dept. of Commun. Sci. and Disord., Northwestern Univ., Tempe, Arizona), Oleg Korzyukov, and Charles R. Larson (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Singers require great effort to avoid vocal distortion at register boundaries, as they are trained to diminish the prominence of register breaks. We examined neural mechanisms underlying voice error detection in singers at their register boundaries. We hypothesized that event-related potentials (ERPs), reflecting brain activity, would be larger if a singer's pitch was unexpectedly shifted toward, rather than away, from their register break. Nine trained singers sustained a musical note for ~3 seconds near their modal register boundaries. As the singers sustained these notes, they heard their voice over headphones shift in pitch (+/- 400 cents, 200 ms) either toward or away from the register boundary. This procedure was repeated for 200 trials. The N1 and P2 ERP amplitudes for three central electrodes (FCz, Cz, Fz) were computed from the EEGs of all participants. Results of a multivariate analysis of variance for shift direction (+400c, -400c) and register (low, high) showed significant differences in N1 and P2 amplitude for direction at the low boundary of modal register, but not the high register boundary. These results may suggest increased neural activity in singers when trying to control the voice when crossing the lower register boundary.

2pSC21. The articulatory tone-bearing unit: Gestural coordination of lexical tone in Thai. Robin P. Karlin and Sam Tilsen (Linguist, Cornell Univ., 103 W Yates St., Ithaca, NY 14850, karlin.robin@gmail.com)

Recently, tones have been analyzed as articulatory gestures that can coordinate with segmental gestures. In this paper, we show that the tone gestures that make up a HL contour tone are differentially coordinated with articulatory gestures in Thai syllables, and that the coordinative patterns are influenced by the segments and moraic structure of the syllables. The autosegmental approach to lexical tone describes tone as a suprasegment that must be associated to some tone-bearing unit (TBU); in Thai, the language of study, the proposed TBU is the mora. Although the autosegmental account largely describes the phonological patterning of tones, it remains unclear how the abstract representation of tone is implemented. An electromagnetic articulograph (EMA) study of four speakers of Thai was conducted to examine the effects of segment type and moraic structure on the coordination of tone gestures. In a HL contour tone, tone gestures behave similarly to consonant gestures, and show patterns of coordination with gestures that correspond to moraic segments. However, there is also a level of coordination between the H and L tone gestures. Based on these results, a model of TBUs is proposed within the Articulatory Phonology framework that incorporates tone-segment coordination as well as tone-tone coordination.

2pSC22. The role of prosody in English sentence disambiguation. Taylor L. Miller (Linguist & Cognit. Sci., Univ. of Delaware, 123 E Main St., Newark, DE 19716, tlmiller@udel.edu)

Only certain ambiguous sentences are perceptually disambiguable. Some researchers argue that this is due to syntactic structure (Lehiste 1973, Price 1991, Kang & Speer 2001), while others argue prosodic structure is responsible (Nespor & Vogel 1986 = N&V, Hirshberg & Avesani 2000). The present study further tests the role of prosodic constituents in sentence disambiguation in English. Target sentences were recorded in disambiguating contexts; twenty subjects listened to the recordings and chose one of two meanings. Following N&V's experimental design with Italian, the meanings of each target structurally corresponded to different syntactic constituents and varied with respect to phonological phrases (ϕ) and intonational phrases (I). The results confirm N&V's Italian findings: listeners are only able to disambiguate sentences with different prosodic constituent structures ($p < 0.05$); those differing in (I) but not (ϕ) have the highest success rate—86% (e.g., [When danger threatens your children]I [call the police]I vs. [When danger threatens]I [your children call the police]I). As reported elsewhere (e.g., Lehiste 1973), we also observed a meaning bias in some cases (e.g., in "Julie ordered some large knife sharpeners," listeners preferred "large [knife sharpeners]" but in "Jill owned some gold fish tanks," they preferred "[goldfish] tanks").

2pSC23. Perceptual isochrony and prominence in spontaneous speech. Tuuli Morrill (Linguist, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA, tmorrill@msu.edu), Laura Dilley (Commun. Sci. and Disord., Michigan State Univ., East Lansing, MI), and Hannah Forsythe (Linguist, Michigan State Univ., East Lansing, MI)

While it has been shown that stressed syllables do not necessarily occur at equal time intervals in speech (Cummins, 2005; Dauer, 1983), listeners frequently perceive stress as occurring regularly, a phenomenon termed perceptual isochrony (Lehiste, 1977). A number of studies have shown that in controlled experimental materials, a perceptually isochronous sequence of stressed syllables generates expectations which affect word segmentation and lexical access in subsequent speech (e.g., Dilley & McAuley, 2008). The present research used the Buckeye Corpus of Conversational Speech (Pitt *et al*., 2007) to address two main questions (1) What acoustic and linguistic factors are associated with the occurrence of perceptual isochrony? and (2) What are the effects of perceptually isochronous speech passages on the placement of prominence in subsequent speech? In particular, we investigate the relationship between perceptual isochrony and lexical items traditionally described as "unstressed" (e.g., grammatical function words), testing whether these words are more likely to be perceived and/or produced as prominent when they are preceded and/or followed by a perceptually isochronous passage. These findings will contribute to our understanding of the relationship between acoustic correlates of phrasal prosody and lexical perception. [Research partially supported by NSF CAREER Award BCS 0874653 to L. Dilley.]

2pSC24. French listeners' processing of prosodic focus. Jui Namjoshi (French, Univ. of Illinois at Urbana-Champaign, 2090 FLB, MC-158, S. Mathews Ave, Urbana, IL 61801, namjosh2@illinois.edu)

Focus in French, typically conveyed by syntax (e.g., clefting) with prosody, can be signaled by prosody alone (contrastive pitch accents on the first syllable of focused constituents, cf. nuclear pitch accents, on the last nonreduced syllable of the Accentual Phrase) (Féry, 2001; Jun & Fougeron, 2000). Do French listeners, like L1-English listeners (Ito & Speer, 2008) use contrastive accents to anticipate upcoming referents? 20 French listeners completed a visual-world eye-tracking experiment. Cross-spliced, amplitude-neutralized stimuli included context (1) and critical (2) sentences in a 2x2 design, with accent on object (nuclear/ contrastive) and person's information status (new/ given) as within-subject variables (see (1)-(2)). Average amplitudes and durations for object words were 67 dB and 0.68 s for contrastive accents, and 63.8 dB and 0.56 s for nuclear accents, respectively. Mixed-effects models showed a significant effect of accent-by-information-status interaction on competitor fixation proportions in the post-disambiguation time window ($p<0.05$). Contrastive accents yielded lower competitor fixation proportions with a given person than with a new person, suggesting that contrastive accents constrain lexical competition in French. (1) Clique sur le macaRON de Marie-Hélène. (2) Puis clique sur le chocoLAT/ CHOcolat de Marie-Hélène/ Jean-Sébastien. (nuclear/contrastive accent, given/new person) '(Then) Click on the macaron/chocolate of Marie-Hélène/Jean-Sébastien.'

2pSC25. Prominence, contrastive focus and information packaging in Ghanaian English discourse. Charlotte F. Lomotey (Texas A&M University-Commerce, 1818D Hunt St., Commerce, TX 75428, cefolatey@yahoo.com)

Contrastive focus refers to the coding of information that is contrary to the presuppositions of the interlocutor. Thus, in everyday speech, speakers employ prominence to mark contrastive focus such that it gives an alternative answer to an explicit or implicit statement provided by the previous discourse or situation (Rooth, 1992), and plays an important role in facilitating language understanding. Even though contrastive focus has been investigated in native varieties of English, there is little or no knowledge of similar studies as far as non-native varieties of English, including that of Ghana, are concerned. The present study investigates how contrastive focus is marked with prosodic prominence in Ghanaian English, and how such a combination creates understanding among users of this variety. To achieve this, data consisting of 61/2 hours of English conversations from 200 Ghanaians were analyzed using both auditory and acoustic means. Results suggest that Ghanaians tend to shift the contrastive focus from the supposed focused syllable onto the last syllable of the utterance, especially when that syllable ends the utterance. Although such tendencies may shift the focus of the utterance, the data suggest that listeners do not seem to have any problem with speakers' packaging of such information.

2pSC26. The representation of tone 3 sandhi in Mandarin: A psycholinguistic study. Yu-Fu Chien and Joan Sereno (Linguist, The Univ. of Kansas, 1407 W 7th St., Apt. 18, Lawrence, KS 66044-6716, whouselefthand@gmail.com)

In Mandarin, tone 3 sandhi is a tonal alternation phenomenon in which a tone 3 syllable changes to a tone 2 syllable when it is followed by another tone 3 syllable. Thus, the initial syllable of Mandarin bisyllabic sandhi words is tone 3 underlyingly but becomes tone 2 on the surface. An auditory-auditory priming lexical decision experiment was conducted to investigate how Mandarin tone 3 sandhi words are processed by Mandarin native listeners. The experiment examined prime-target pairs, with monosyllabic primes and bisyllabic Mandarin tone 3 sandhi targets. Each tone sandhi target word was preceded by one of three corresponding monosyllabic primes: a tone 2 prime (Surface-Tone overlap) (chu2-chu3li3), a tone 3 prime (Underlying-Tone overlap) (chu3-chu3li3), or a control prime (Baseline condition) (chu1-chu3li3). In order to assess the contribution of frequency of occurrence, 15 High Frequency and 15 Low Frequency sandhi target words were used. Thirty native speakers of Mandarin participated. Results showed that tone 3 sandhi targets elicited significantly stronger facilitation effects in the Underlying-Tone condition than in the Surface-Tone condition, with little effect of frequency of occurrence. The data will be discussed in terms of lexical access and the nature of the representation of Mandarin words.

2pSC27. Perception of sound symbolism in mimetic stimuli: The voicing contrast in Japanese and English. Kotoko N. Grass (Linguist, Univ. of Kansas, 9953 Larsen St., Overland Park, KS 66214, nakata.k@ku.edu) and Joan Sereno (Linguist, Univ. of Kansas, Lawrence, KS)

Sound symbolism is a concept in which the sound of a word and the meaning of the word are systematically related. The current study investigated whether the voicing contrast between voiced /d, g, z/ and voiceless /t, k, s/ consonants systematically affects categorization of Japanese mimetic stimuli along a number of perceptual and evaluative dimensions. For the nonword stimuli, voicing of consonants was also manipulated, creating a continuum from voiced to voiceless endpoints (e.g., [gede] to [kete]), in order to examine the categorical nature of the perception. Both Japanese native speakers and English native speakers, who had no knowledge of Japanese, were examined. Stimuli were evaluated on size (big–small) and shape (round–spiky) dimensions as well as two evaluative dimensions (good–bad, graceful–clumsy). In the current study, both Japanese and English listeners associated voiced sounds with largeness, badness, and clumsiness and voiceless sounds with smallness, goodness, and gracefulness. For the shape dimension, however, English and Japanese listeners showed contrastive categorization, with English speakers associating voiced stops with roundness and Japanese listeners associating voiced stops with spikiness. Interestingly, sound symbolism was very categorical in nature. Implications of the current data for theories of sound symbolism will be discussed.

Session 2pUW

Underwater Acoustics: Propagation and Scattering

Megan S. Ballard, Chair
Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78758

Contributed Papers

1:00

2pUW1. Low frequency propagation experiments in Currituck Sound. Richard D. Costley (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, dan.costley@usace.army.mil), Kent K. Hathaway (Coastal & Hydraulics Lab., US Army Engineer Res. & Development Ctr., DC, NC), Andrew McNeese, Thomas G. Muir (Appl. Res. Lab., Univ. of Texas at Austin, Austin, TX), Eric Smith (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, Texas), and Luis De Jesus Diaz (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS)

In water depths on the order of a wavelength, sound propagates with considerable involvement of the bottom, whose velocities and attenuation vary with depth into the sediment. In order to study propagation in these types of environments, experiments were conducted in Currituck Sound on the Outer Banks of North Carolina using a Combustive Sound Source (CSS) and bottom mounted hydrophones and geophones as receivers. The CSS was deployed at a depth of approximately 1 meter and generated transient signals, several wavelengths long, at frequencies around 300 Hz. The results are used to determine transmission loss in water depths of approximately 3 meters, as well as to examine the generation and propagation of Sholte type interface waves. The measurements are compared to numerical models generated with a two-dimensional finite-element code. [Work supported by the U.S. Army Engineer Research and Development Center. Permission to publish was granted by Director, Geotechnical & Structures Laboratory.]

1:15

2pUW2. Three-dimensional acoustic propagation effect in subaqueous sand dune field. Andrea Y. Chang, Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, yychang@ntu.edu.tw), Linus Y. Chiu (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), Emily Liu (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), Ching-Sang Chiu, and Davis B. Reeder (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA)

Very large subaqueous sand dunes are discovered on the upper continental slope of the northern South China Sea in water depth of 160–600 m, which composed of fine to medium sand. The amplitude and the crest-to-crest wavelength of sand dunes are about 5–15 m and 200–400 m, respectively. This topographic feature could causes strong acoustic scattering, mode coupling, and out-of- plane propagation effects, which consequently result in sound energy redistribution within ocean waveguide. This research focus on the three-dimensional propagation effects (e.g., horizontal refraction) induced by the sand dunes in the South China Sea, which are expected as the angle of propagation relative to the bedform crests decreases. The three-dimensional propagation effects are studied by numerical modeling and model-data comparison. For numerical modeling, the *in-situ* topographic data of subaqueous sand dune and sound speed profiles were inputted to calculate the acoustic fields, which were further decomposed into mode fields to show the modal horizontal refraction effects. The modeling results were manifested by data observations. [This work is sponsored by the Ministry of Science and Technology of Taiwan.]

1:30

2pUW3. Results from a scale model acoustic propagation experiment over a translationally invariant wedge. Jason D. Sagers (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

A 1:7500 scale underwater acoustic propagation experiment was conducted in a laboratory tank to investigate three-dimensional (3D) propagation effects, with the objective of providing benchmark quality data for comparison with numerical models. A computer controlled positioning system accurately moves the receiving hydrophone in 3D space while a stationary source hydrophone emits band-limited pulse waveforms between 200 kHz and 1 MHz. The received time series can be post-processed to estimate travel time, transmission loss, and vertical and horizontal arrival angle. Experimental results are shown for a 1.22×2.13 m bathymetric part possessing both a flat bottom bathymetry and a translationally invariant wedge with a 10° slope. Comparisons between the experimental data and numerical models are also shown. [Work supported by ONR.]

1:45

2pUW4. Numerical modeling of measurements from an underwater scale-model tank experiment. Megan S. Ballard and Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

Scale-model tank experiments are beneficial because they offer a controlled environment in which to make underwater acoustic propagation measurements, which is helpful when comparing measured data to calculations from numerical propagation models. However, to produce agreement with the measured data, experimental details must be carefully included in the model. For example, the frequency-dependent transmitting and receiving sensitivity and vertical directionality of both hydrophones must be included. In addition, although it is possible to measure the geometry of the tank experiment, including water depth and source and receiver positions, positional uncertainty exists due to the finite resolution of the measurements. The propagated waveforms from the experiment can be used to resolve these parameters using inversion techniques. In this talk, model-data comparisons of measurements made in a 1:7500 scale experiment are presented. The steps taken to produce agreement between the measured and modeled data are discussed in detail for both range-independent and range-dependent configurations.

2:00

2pUW5. A normal mode inner product to account for acoustic propagation over horizontally variable bathymetry. Charles E. White, Cathy Ann Clark (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, charlie.e.white@navy.mil), Gopu Potty, and James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

This talk will consider the conversion of normal mode functions over local variations in bathymetry. Mode conversions are accomplished through an inner product, which enables the modes compromising the field at each range-dependent step to be written as a function of those in the preceding step. The efficiency of the method results from maintaining a stable number

of modes throughout the calculation of the acoustic field. A verification of the inner product is presented by comparing results from its implementation in a simple mode model to that of a closed-form solution for the acoustic wedge environment. A solution to the more general problem of variable bottom slope, which involves a decomposition of bathymetric profiles into a sequence of wedge environments, will also be discussed. The overall goal of this research is the development and implementation of a rigorous shallow water acoustic propagation solution which executes in a time window to support tactical applications.

2:15

2pUW6. An assessment of the effective density fluid model for backscattering from rough poroelastic interfaces. Anthony L. Bonomo, Nicholas P. Chotiros, and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

The effective density fluid model (EDFM) was developed to approximate the behavior of sediments governed by Biot's theory of poroelasticity. Previously, it has been shown that the EDFM predicts reflection coefficients and backscattering strengths that are in close agreement with those of the full Biot model for the case of a homogeneous poroelastic half-space. However, it has not yet been determined to what extent the EDFM can be used in place of the full Biot model for other cases. In this work, the finite element method is used to compare the backscattering strengths predicted using the EDFM with the predictions of the full Biot model for three cases: a homogeneous poroelastic half-space with a rough interface, a poroelastic layer overlying an elastic half-space with both interfaces rough, and an inhomogeneous poroelastic half-space consisting of a shear modulus gradient with a rough interface. [Work supported by ONR, Ocean Acoustics.]

2:30

2pUW7. Scattering by randomly rough surfaces. I. Analysis of slope approximations. Patrick J. Welton (Appl. Res. Lab., The Univ. of Texas at Austin, 1678 Amarelle St., Thousand Oaks, CA 91320-5971, patrickwelton@verizon.net)

Progress in numerical methods now allows scattering in two dimensions to be computed without resort to approximations. However, scattering by three-dimensional random surfaces is still beyond the reach of current numerical techniques. Within the restriction of the Kirchhoff approximation (single scattering) some common approximations used to predict scattering by randomly rough surfaces will be examined. In this paper, two widely used approximate treatments for the surface slopes will be evaluated and compared to the exact slope treatment.

2:45–3:00 Break

3:00

2pUW8. Scattering by randomly rough surfaces. II. Spatial spectra approximations. Patrick J. Welton (Appl. Res. Lab., The Univ. of Texas at Austin, 1678 Amarelle St., Thousand Oaks, CA 91320-5971, patrickwelton@verizon.net)

The spatial spectrum describing a randomly rough surface is crucial to the theoretical analysis of the scattering behavior of the surface. Most of the models assume that the surface displacements are a zero-mean process. It is shown that a zero-mean process requires that the spatial spectrum vanish when the wavenumber is zero. Many of the spatial spectra models used in the literature do not meet this requirement. The impact of the zero-mean requirement on scattering predictions will be discussed, and some spectra models that meet the requirement will be presented.

3:15

2pUW9. Scattering by randomly rough surfaces. III. Phase approximations. Patrick J. Welton (Appl. Res. Lab., The Univ. of Texas at Austin, 1678 Amarelle St., Thousand Oaks, CA 91320-5971, patrickwelton@verizon.net)

In the limit as the roughness vanishes, the solution for the pressure scattered by a rough surface of infinite extent should reduce to the image solution. Approximate image solutions for an infinite, pressure-release plane surface are studied for an omnidirectional source using the 2nd, 3rd, and 4th order phase approximations. The results are compared to the exact image solution to examine the effects of the phase approximations. The result based on the 2nd order (Fresnel phase) approximation reproduces the image solution for all geometries. Surprisingly, the results for the 3rd and 4th order phase approximations are never better than the Fresnel result, and are substantially worse for most geometries. This anomalous behavior is investigated and the cause is found to be the multiple stationary phase points produced by the 3rd and 4th order phase approximations.

3:30

2pUW10. Role of binding energy (edge-to-face contact of mineral platelets) in the acoustical properties of oceanic mud sediments. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net) and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

A theory for mud sediments presumes a card-house model, where the platelets arrange themselves in a highly porous configuration; electrostatic forces prevent face-to-face contacts. The primary type of contact is where the edge of one platelet touches a face of another. Why such is not also prevented by electrostatic forces is because of van der Waals (vdW) forces between the molecular structures within the two platelets. A quantitative assessment is given of such forces, taking into account the atomic composition and crystalline structure of the platelets, proceeding from the London theory of interaction between non-polar molecules. Double-integration over both platelets leads to a quantitative and simple prediction for the potential energy of vdW interaction as a function of the separation distance, edge-from-face. At moderate nanoscale distances, the resulting force is attractive and is much larger than the electrostatic repulsion force. But, at very close (touching) distances, the intermolecular force becomes also repulsive, so that there is a minimum potential energy, which is identified as the binding energy. This finite binding energy, given a finite environmental temperature, leads to some statistical mechanical theoretical implications. Among the acoustical implications is a relaxation mechanism for the attenuation of acoustic waves propagating through mud.

3:45

2pUW11. Near bottom self-calibrated measurement of normal reflection coefficients by an integrated deep-towed camera/acoustical system. Linus Chiu, Chau-Chang Wang, Hsin-Hung Chen (Inst. of Undersea Technol., National Sun Yat-sen Univ., No. 70, Lienhai Rd., Kaohsiung 80424, Taiwan, linus@mail.nsysu.edu.tw), Andrea Y. Chang (Asia-Pacific Ocean Res. Ctr., National Sun Yat-sen Univ., Kaohsiung, Taiwan), and Chung-Ray Chu (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

Normal incidence echo data (bottom reflection) can provide acoustic reflectivity estimates used to predict sediment properties with using seabed sediment models. Accuracy of normal reflection coefficient measurement thus become very significant to the bottom inversion result. A deep-towed camera platform with acoustical recording system, developed by the Institution of Undersea Technology, National Sun Yat-sen University, Taiwan, is capable of photographically surveying the seafloor in near scope and acquiring sound data. The real time data transference, including photography (optics) and reflection measurement (acoustics) can be implemented in the same site simultaneously. The deep-towed camera near the bottom was used in several experiments in the southwestern sea off Taiwan in 2014 to acquire acoustic LFM signal sent by surface shipboard source as incident signal as well as the seafloor reflections at frequency bands within 4–6 kHz. The error produced by compensating the roll-off of altitude of vehicle (propagation loss) can be eliminated, which is considered as near bottom self-calibrated measurement for normal reflection coefficient. The collected reflection coefficients were used to inverting the sediment properties with using the Effective Density Fluid model (EDFM), manifested by the coring and camera images. [This work is sponsored by the Ministry of Science and Technology of Taiwan.]

4:00

2pUW12. Backscattering from an obstacle immersed in an oceanic waveguide covered with ice. Natalie S. Grigorieva (St. Petersburg State Electrotech. Univ., 5 Prof. Popova Str., St. Petersburg 197376, Russian Federation, nsgrig@natalie.spb.su), Daria A. Mikhaylova, and Dmitriy B. Ostrovskiy (JSC "Concern Oceanpribor", St. Petersburg, Russian Federation)

The presentation describes the theory and implementation issues of modeling of the backscattering from an obstacle immersed in a homogeneous, range-independent waveguide covered with ice. An obstacle is assumed to be spherical, rigid or fluid body. A bottom of the waveguide and an ice cover are fluid, attenuating half-space. The properties of an ice cover and a scatterer may coincide. To calculate the scattering coefficients of a sphere [R. M. Hackman *et al*., J. Acoust. Soc. Am. 84, 1813–1825 (1988)], the normal mode evaluation is applied. A number of normal modes forming the backscattering field is determined by a given directivity of the source. The obtained analytical expression for the backscattered field is applied to evaluate its dependence on source frequency, depth of a water layer, bottom and ice properties, and distance between the source and obstacle. Two cases are analyzed and compared: when the upper boundary of a waveguide is sound-soft and when a water layer is covered with ice. Computational results are obtained in a wide frequency range 8–12 kHz for conditions of a shallow water testing area. [Work supported by Russian Ministry of Educ. and Sci., Grant 02.G25.31.0058.]

4:15

2pUW13. Emergence of striation patterns in acoustic signals reflected from dynamic surface waves. Youngmin Choo, Woojae Seong (Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul, Seoul 151 - 744, South Korea, sks655@snu.ac.kr), and Heechun Song (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA)

A striation pattern can emerge in high-frequency acoustic signals interacting with dynamic surface waves. The striation pattern is analyzed using a ray tracing algorithm for both a sinusoidal and a rough surface. With a source or receiver close to the surface, it is found that part of the surface on either side of the specular reflection point can be illuminated by rays, resulting in time-varying later arrivals in channel impulse response that form the striation pattern. In contrast to wave focusing associated with surface wave crests, the striation occurs due to reflection off convex sections around troughs. Simulations with a sinusoidal surface show both an upward (advancing) and downward (retreating) striation patterns that depend on the surface-wave traveling direction and the location of the illuminated area. In addition, the striation length is determined mainly by the depth of the source or receiver, whichever is closer in range to the illuminated region. Even with a rough surface, the striation emerges in both directions. However, broadband (7–13 kHz) simulations in shallow water indicate that the longer striation in one direction is likely pronounced against a quiet noise background, as observed from at-sea experimental data. The simulation is extended for various surface wave spectra and it shows consistent patterns.

TUESDAY EVENING, 28 OCTOBER 2014 — 8:00 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. On Tuesday the meetings will begin at 8:00 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m. On Thursday evening, the meetings will begin at 7:30 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.)	Santa Fe
Acoustical Oceanography	Indiana G
Architectural Acoustics	Marriott 7/8
Physical Acoustics	Indiana C/D
Speech Communication	Marriott 3/4
Structural Acoustics and Vibration	Marriott 1/2

Session 3aAA

Architectural Acoustics and Noise: Design and Performance of Office Workspaces in High Performance Buildings

Kenneth P. Roy, Chair
Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604

Chair's Introduction—8:20

Invited Papers

8:25

3aAA1. Architecture and acoustics … form and function—What comes 1st? Kenneth Roy (Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604, kproy@armstrong.com)

When I first studied architecture, it was expected that "form fits function" was pretty much a mantra to design. But is that the case today or has it ever been when acoustics are concerned? Numerous post occupancy studies of worker satisfaction with office IEQ indicate that things are not as they should be. And, as a matter of fact, high performance green buildings seem to fair much worse than normal office buildings when acoustic quality is considered. So what are we doing wrong—maybe the Gensler Workplace Study and other related studies could shed light on what is wrong, and how we might think differently about office design. From an acousticians viewpoint it's all about "acoustic comfort" meaning the right amount of intelligibility, privacy, and distraction for the specific work function. Times change and work functions change, so maybe we should be looking for a new mantra … like "function drives form." We may also want to consider that office space may need to include a "collaboration zone" where teaming takes place, a "focus zone" where concentrated thought can take place, and a "privacy zone" where confidential discussions can take place. Each of these requires different architecture and acoustic performance.

8:45

3aAA2. Acoustics in collaborative open office environments. John J. LoVerde, Samantha Rawlings, and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Historically, acoustical design for open office environments focuses on creating workspaces that maximize speech privacy and minimize aural distractions. Hallmark elements of the traditional open office environment include barriers, sound-absorptive surfaces, and consideration of workspace orientation, size, and background sound level. In recent years, development of "collaborative" office environments has been desired, which creates an open work setting, allowing immediate visual and aural communication between team members. This results in reducing the size of workstations, lowering barriers, and reducing distance between occupants. Additionally, group meeting areas have also become more open, with the popularization of "huddle zones" where small groups hold meetings in an open space adjacent to workstations rather than within enclosed conference rooms. Historically, this type of office environment would have poor acoustical function, with limited speech privacy between workstations and minimal attenuation of distracting noises, leading to occupant complaints. However, these collaborative open office environments function satisfactorily and seem to be preferred by occupants and employers alike. This paper investigates the physical acoustical parameters of collaborative open office spaces.

9:05

3aAA3. Lessons learned in reconciling high performance building design with acoustical comfort. Valerie Smith and Ethan Salter (Charles M. Salter Assoc., 130 Sutter St., Fl. 5, San Francisco, CA 94104, valerie.smith@cmsalter.com)

In today's diverse workplace, "the one size fits all" approach to office design is becoming less prevalent. The indoor environmental quality of the workplace is important to owners and occupants. Architects are developing innovative ways to encourage interaction and collaboration while also increasing productivity. Many of these ideas are at odds with the traditional acoustical approaches used for office buildings. Employees are asking for, and designers are incorporating, amenities such as kitchens, game rooms, and collaboration spaces into offices. Architects and end users are becoming increasingly aware of acoustics in their environment. The U.S. General Services Administration (GSA) research documents, as well as those from other sources, discusses the importance of acoustics in the workplace. Private companies are also creating acoustical standards documents for use in design of new facilities. As more buildings strive to achieve sustainable benchmarks (whether corporate, common green building rating systems such as LEED, or code-required) the understanding of the need for acoustical items (such as sound isolation, speech privacy, and background noise) also become critical. The challenge is how to reconcile sustainable goals with acoustical features. This presentation discusses several of the approaches that our firm has recently used in today's modern office environment.

Contributed Papers

9:25

3aAA4. I can see clearly now, but can I also hear clearly now too? Patricia Scanlon, Richard Ranft, and Stephen Lindsey (Longman Lindsey, 1410 Broadway, Ste. 508, New York, NY 10018, patricias@longmanlindsey.com)

The trend in corporate workplace has been away from closed plan gypsum board offices to open plan workstations and offices with glass fronts, sliding doors, and clearstories or glass fins in the wall between offices. These designs are often a kit of parts supplied by manufacturers, who offer minimal information on the sound transmission that will be achieved in practice. This results in end users who are often misled into believing they will enjoy a certain level of speech privacy in their offices. Our presentation will discuss the journey from benchmarking the NIC rating of an existing office construction, reviewing the STC ratings for various glass front options, evaluating details including door frame, door seals, intersection between office demising walls and front partition systems. We will then present how this information is transferred to the client, allowing them to make an informed decision on the construction requirements for their new space. We will highlight the difference in acoustical environment between what one might expect from reading manufacturer's literature, and what is typically achieved in practice.

9:40

3aAA5. Acoustics in an office building. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

New building techniques tend to make better use of materials, temperature, and energy, besides costs. A building company had to plan the adaptation of a very old building with the purpose to install private offices of different sizes in each floor, in order to take advantage of a large solid construction, reducing building time, total weight, etc., while at the same time fulfilling new requirements related with comfort, general quality, functionality, and economy. Among several other topics, sound and vibrations had to be considered during the process, including noise control and speech privacy, because a combination of private rooms and open plan offices were needed, as well as limiting environmental vibrations. Aspects such as the use of light weight materials and the installation of many climate conditioning systems were needed, which were dealt along the project in the search for a long lasting and low maintenance costs construction.

9:55–10:10 Break

Invited Papers

10:10

3aAA6. A case history in architectural acoustics: Security, acoustics, the protection of personally identifiable information (PII), and accessibility for the disabled. Donna A. Ellis (The Div. of Architecture and Eng., The Social Security Administration, 415 Riggs Ave., Severna Park, MD 21146, Donna.a.ellis@ssa.gov)

This paper discusses the re-design of a field office to enhance the protection of Personally Identifiable Information (PII), physical security, and accessibility for the disabled at the Social Security Administration (SSA) field office in Roxbury, MA. The study, and its results can be used at federal, civil, and private facilities where transaction window type public interviews occur. To protect the public and its staff, the SSA has mandated heightened security requirements in all field offices. The increased security measures include: Installation of barrier walls to provide separation between the public and private zones; maximized lines of sight, and increased speech privacy for the protection of PII. This paper discusses the use of the Speech Transmission Index (STI) measurement method used to determine the post construction intelligibility of speech through the transaction window, the acoustical design of the windows and their surrounding area, how appropriate acoustic design helps safeguard personal and sensitive information so that it may be securely communicated verbally, as well as improved access for the disabled community, especially the hearing impaired.

10:30

3aAA7. High performance medical clinics: Evaluation of speech privacy in open-plan offices and examination rooms. Steve Pettyjohn (The Acoust. & Vib. Group, Inc., 5765 9th Ave., Sacramento, CA CA, spettyjohn@acousticsandvibration.com)

Speech privacy evaluations of open plan doctors' offices and examination rooms were done at two clinics. One was in Las Vegas and the other in El Dorado Hills. The building were designed to put doctors closer to patients and for a cost savings. ASTM E1130, ASTM E336, and NRC guidelines were used to evaluate these spaces. For E1130, sound is produced at the source location with calibrated speakers, then measurements are made at receiver positions. The speaker faces the receiver. Only open plan furniture separated the source from the receiver. The examination rooms used partial height walls with a single layer of gypsum board on each face. Standard doors without seals were used. CAC 40 rated ceiling tile were installed. The cubicle furniture included sound absorption and was 42 to 60 in. tall. The Privacy Index was quite low, ranging from 30 to 66%. The NIC rating of the walls without doors ranged from 38 to 39, giving PI ratings of 83 to 84%. With a door, the NIC rating was 30 to 31 with PI ratings of 72. These results do not meet the requirements of the Facility Guideline Institute or ANSI 12 Working Group 44.

10:50

3aAA8. Exploring the impacts of consistency in sound masking. Niklas Moeller and Ric Doedens (K.R. Moeller Assoc. Ltd., 1050 Pachino Court, Burlington, ON L7L 6B9, Canada, rdoedens@logison.com)

Electronic sound masking systems control the noise side of the signal-to-noise ratio in interior environments. Their effectiveness relates directly to how consistently the specified masking curve is achieved. Current system specifications generally allow a relatively wide range in performance, in large part reflecting expectations set by legacy technologies. This session presents a case study of sound masking measurements and speech intelligibility calculations conducted in office spaces. These are used as a foundation to discuss the impacts of local inconsistencies in the masking sound and to begin a discussion of appropriate performance requirements for masking systems.

11:10

3aAA9. Evaluating the effect of prominent tones in noise on human task performance. Joonhee Lee and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, joonhee.lee@huskers.unl.edu)

Current noise guidelines for the acoustic design of offices generally specify limits on loudness and sometimes spectral shape, but do not typically address the presence of tones in noise as may be generated by building services equipment. Numerous previous studies indicate that the presence of prominent tones is a significant source of deteriorating indoor environmental quality. Results on how prominent tones in background noise affect human task performance, though, are less conclusive. This paper presents results from recent studies at Nebraska on how tones in noise may influence task performance in a controlled office-like environment. Participants were asked to complete digit span tasks as a measure of working memory capacity, while exposed to assorted noise signals with tones at varying frequencies and tonality levels. Data on the percent correct and reaction time in which participants responded to the task are analyzed statistically. The results can provide guidance for setting limits on the tonality levels in offices and other spaces in which building users must be task-productive.

Contributed Paper

11:30

3aAA10. Optimal design of multi-layer microperforated sound absorbers. Nicholas Kim, Yutong Xue, and J. S. Bolton (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 S. Russell St., West Lafayette, IN, kim505@purdue.edu)

Microperforated polymer films can offer and effective solution when it is desired to design fiber-free sound absorption systems. The acoustic performance of the film is determined by hole size and shape, by the surface porosity, by the mass per unit area of the film, and by the depth of the backing air layer. Single sheets can provide good absorption over a one of two octave range, but if absorption over a broader range is desired, it is necessary to use multilayer treatments. Here the design of a multilayer sound absorption system is described, where the film is considered to have a finite mass per unit area and also to have conical perforations. It will be shown that it is possible to design compact absorbers that yield good performance over the whole speech interference range. In the course of the optimization it has been found that there is a tradeoff between cone angle and surface porosity. The design of lightweight, multilayer functional absorbers will also be described, and it will be shown, for example, that it is possible to design systems that simultaneously possess good sound absorption and barrier characteristics.

Session 3aAB

Animal Bioacoustics: Predator–Prey Relationships

Simone Baumann-Pickering, Cochair
Scripps Institution of Oceanography, University of California San Diego, 9500 Gilman Dr, La Jolla, CA 92093

Ana Sirovic, Cochair
Scripps Institution of Oceanography, 9500 Gilman Drive MC 0205, La Jolla, CA 92093-0205

Chair's Introduction—8:25

Invited Papers

8:30

3aAB1. Breaking the acoustical code of ants: The social parasite's pathway. Francesca Barbero, Luca P. Casacci, Emilio Balletto, and Simona Bonelli (Life Sci. and Systems Biology, Univ. of Turin, Via Accademia Albertina 13, Turin 10123, Italy, francesca.barbero@unito.it)

Ant colonies represent a well-protected and stable environment (temperature, humidity) where essential resources are stored (e.g., the ants themselves, their brood, stored food). To maintain their social organization, ants use a variety of communication channels, such as the exchange of chemical and tactile signals, as well as caste specific stridulations (Casacci *et al.* 2013 Current Biology 23, 323–327). By intercepting and manipulating their host's communication code, about 10,000 arthropod species live as parasites and exploit ant nests. Here, we review results of our studies on *Maculinea* butterflies, a group of social parasites which mimic the stridulations produced by their host ants to promote (i) their retrieval into the colony (adoption: Sala *et al.* 2014, PLoS ONE 9(4), e94341), (ii) their survival inside the nest/brood chambers (integration: Barbero *et al.* 2009 J. Exp. Biol. 218, 4084–4090), or (iii) their achievement of the highest possible social status within the colony's hierarchy (full integration: Barbero *et al.* 2009, Science 323, 782–785). We strongly believe that the study of acoustic communication in ants will bring about significant advances in our understanding of the complex mechanisms underlying the origin, evolution, and stabilization of many host–parasite relationships.

8:50

3aAB2. How nestling birds acoustically monitor parents and predators. Andrew G. Horn and Martha L. Leonard (Biology, Dalhousie Univ., Life Sci. Ctr., 1355 Oxford St., PO Box 15000, Halifax, NS B3H 4R2, Canada, aghorn@dal.ca)

The likelihood that nestling songbirds survive until leaving the nest depends largely on how well they are fed by parents and how well they escape detection by predators. Both factors in turn are affected by the nestling's begging display, a combination of gaping, posturing, and calling that stimulates feedings from parents but can also attract nest predators. If nestlings are to be fed without being eaten themselves, they must beg readily to parents but avoid begging when predators are near the nest. Here we describe experiments to determine how nestling tree swallows, *Tachycineta bicolor*, use acoustic cues to detect the arrival of parents with food and to monitor the presence of predators, in order to beg optimally relative to their need for food. We also discuss how their assessments vary in relation to two constraints: their own poor perceptual abilities and ambient background noise. Together with similar work on other species, our research suggests that acoustically conveyed information on predation risk has been an important selective force on parent-offspring communication. More generally, how birds acoustically monitor their environment to avoid predation is an increasingly productive area of research.

9:10

3aAB3. Acoustic preferences of frog-biting midges in response to intra- and inter-specific signal variation. Ximena Bernal (Dept. of Biological Sci., Purdue Univ., 915 W. State St., West Lafayette, IN 47906, xbernal@purdue.edu)

Eavesdropping predators and parasites intercept mating signals emitted by their prey and host gaining information that increases the effectiveness of their attack. This kind of interspecific eavesdropping is widespread across taxonomic groups and sensory modalities. In this study, sound traps and a sound imaging device system were used to investigate the acoustic preferences of frog-biting midges, *Corethrella* spp. (Corethrellidae). In these midges, females use the advertisement call produced by male frogs to localize them and obtained a blood meal. As in mosquitoes (Culicidae), a closely related family, female midges require blood from their host for egg production. The acoustic preferences of the midges were examined in the wild in response to intra- and interspecific call variation. When responding to call variation in túngara frogs (*Engystomops pustulosus*), frogs producing vocalizations with higher call complexity and call rate were preferentially attacked by the midges. Túngara frog calls were also preferred by frog-biting midges over the calls produced by a sympatric frog of similar size, the hourglass frog (*Dendrosophus ebbracatus*). The role of call site selection in multi-species aggregations is explored in relation to the responses of frog-biting midges. In addition, the use of acoustic traps and sound imaging devices to investigate eavesdropper-victim interactions are discussed.

Contributed Papers

9:30

3aAB4. Foraging among acoustic clutter and competition: Vocal behavior of paired big brown bats. Michaela Warnecke (Psychol. and Brain Sci., The Johns Hopkins Univ., 3400 N Charles St., Baltimore, MD 21218, michaela.warnecke@jhu.edu), Chen Chiu, Wei Xian (Psychol. and Brain Sci., The Johns Hopkins Univ., Baltimore, MD), Clement Cechetto (AGROSUP, Inst. Nationale Superieur des Sci. Agronomique, Dijon, France), and Cynthia F. Moss (Psychol. and Brain Sci., The Johns Hopkins Univ., Baltimore, MD)

In their natural environment, big brown bats forage for small insects in open spaces, as well as in the presence of acoustic clutter. While searching and hunting for prey, these bats experience sonar interference not only from densely cluttered environments, but also through calls from other conspecifics foraging close-by. Previous work has shown that when two bats fly in a relatively open environment, one of them may go silent for extended periods of time (Chiu *et al.* 2008), which may serve to minimize such sonar interference between conspecifics. Additionally, big brown bats have been shown to adjust frequency characteristics of their vocalizations to avoid acoustic interference from conspecifics (Chiu *et al.*, 2009). It remains an open question, however, in what way environmental clutter and the presence of conspecifics influence the bat's call behavior. By recording multichannel audio and video data of bats engaged in insect capture in an open and a cluttered space, we quantified the bats' vocal behavior. Bats were flown individually and in pairs in an open and cluttered room, and the results of this study shed light on the strategies animals employ to negotiate a complex and dynamic environment.

9:45

3aAB5. Sensory escape from a predator–prey arms race: Low amplitude biosonar beats moth hearing. Holger R. Goerlitz (Acoust. and Functional Ecology, Max Planck Inst. for Ornithology, Eberhard-Gwinner-Str, Seewiesen 82319, Germany, hgoerlitz@orn.mpg.de), Hannah M. ter Hofstede (Biological Sci., Dartmouth College, Hanover, NH), Matt Zeale, Gareth Jones, and Marc W. Holderied (School of Biological Sci., Univ. of Bristol, Bristol, United Kingdom)

Ultrasound-sensitive ears evolved in many nocturnal insects, including some moths, to detect bat echolocation calls and evade capture. Although there is evidence that some bats emit echolocation calls that are inconspicuous to eared moths, it is difficult to determine whether this was an adaptation to moth hearing or originally evolved for a different purpose. Here we present the first example of an echolocation counterstrategy to overcome prey hearing at the cost of reduced detection distance, providing an example of a predator outcompeting its prey despite the life-dinner-principle. Aerial-hawking bats generally emit high-amplitude echolocation calls to maximize detection range. Using comparative acoustic flight-path tracking of free-flying bats, we show that the barbastelle, *Barbastella barbastellus*, emits calls that are 10 to 100 times lower in amplitude than those of other aerial hawking bats. Model calculations demonstrate that only bats emitting such low-amplitude calls hear moth echoes before their calls are conspicuous to moths. We confirm that the barbastelle remains undetected by moths until close and preys mainly on eared moths, using moth neurophysiology in the field and fecal DNA analysis. This adaptive stealth echolocation allows the barbastelle to access food resources that are difficult to catch for high-intensity bats.

10:00–10:20 Break

Invited Papers

10:20

3aAB6. Cues, creaks, and decoys: Using underwater acoustics to study sperm whale interactions with the Alaskan black cod longline fishery. Aaron Thode (SIO, UCSD, 9500 Gilman Dr, MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Janice Straley (Univ. of Alaska, Southeast, Sitka, AK), Lauren Wild (Sitka Sound Sci. Ctr., Sitka, AK), Jit Sarkar (SIO, UCSD, La Jolla, CA), Victoria O'Connell (Sitka Sound Sci. Ctr., Sitka, AK), and Dan Falvey (Alaska Longline Fisherman's Assoc., Sitka, AK)

For decades off SE Alaska, sperm whales have located longlining fishing vessels and removed, or "depredated," black cod from the hauls. In 2004, the Southeast Alaska Sperm Whale Avoidance Project (SEASWAP) began deploying passive acoustic recorders on longline fishing gear in order to identify acoustic cues that may alert whales to fishing activity. It was found that when hauling, longlining vessels generate distinctive cavitation sounds, which served to attract whales to the haul site. The combined use of underwater recorders and video cameras also confirmed that sperm whales generated "creak/buzz" sounds while depredating, even under good visual conditions. By deploying recorders with federal sablefish surveys over two years, a high correlation was found between sperm whale creak rate detections and visual evidence for depredation. Thus passive acoustics is now being used as a low-cost, remote sensing method to quantify depredation activity in the presence and absence of various deterrents. Two recent developments will be discussed in detail: the development and field testing of acoustic "decoys" as a potential means of attracting animals away from locations of actual fishing activity, and the use of "TadPro" cameras to provide combined visual and acoustic observations of longline deployments. [Work supported by NPRB, NOAA, and BBC.]

10:40

3aAB7. Follow the food: Effects of fish and zooplankton on the behavioral ecology of baleen whales. Joseph Warren (Stony Brook Univ., 239 Montauk Hwy, Southampton, NY 11968, joe.warren@stonybrook.edu), Susan E. Parks (Dept. of Biology, Syracuse Univ., Syracuse, NY), Heidi Pearson (Univ. of Alaska, Southeast, Juneau, AK), and Kylie Owen (Univ. of Queensland, Gatton, QLD, Australia)

Active acoustics were used to collect information on the type, distribution, and abundance of baleen whale prey species such as zooplankton and fish at fine spatial (sub-meter) and temporal (sub-minute) scales. Unlike other prey measurement methods, scientific echosounder surveys provide prey data at a resolution similar to what a predator would detect in order to efficiently forage. Data from

several studies around the world shows that differences in prey type or distribution result in distinctly different baleen whale foraging behaviors. Humpback whales in coastal waters of Australia altered their foraging pattern depending on the presence and abundance of baitfish or krill. In Southeast Alaska, humpback whales foraged cooperatively or independently depending on prey type and abundance. Humpback whales in the Northwest Atlantic with multiple prey species available foraged on an energetically costly (and presumably rewarding) species. The vertical and horizontal movements of North Atlantic right whales in Cape Cod Bay were strongly correlated with very dense aggregations of copepods. In all of these cases, active acoustics were used to estimate numerical densities of the prey, which provides quantitative information about the energy resource available to foraging animals.

Contributed Papers

11:00

3aAB8. Association of low oxygen waters with the depths of acoustic scattering layers in the Gulf of California and implications for the success of Humboldt squid (*Dosidicus gigas*). David Cade (BioSci., Stanford Univ., 120 Oceanview Boulevard, Pacific Grove, CA 93950, davecade@stanford.edu) and Kelly J. Benoit-Bird (CEOAS, Oregon State Univ., Corvallis, OR)

The ecology in the Gulf of California has undergone dramatic changes over the past century as Humboldt squid (*Dosidicus gigas*) have become a dominant predator in the region. The vertical overlap between acoustic scattering layers, which consist of small pelagic organisms that make up the bulk of *D. gigas* prey, and regions of severe hypoxia have led to a hypothesis linking the shoaling of oxygen minimum zones over the past few decades to compression of acoustic scattering layers, which in turn would promote the success of *D. gigas*. We tested this hypothesis by looking for links between specific oxygen values and acoustic scattering layer boundaries. We applied an automatic layer detection algorithm to shipboard echosounder data from four cruises in the Gulf of California. We then used CTD data and a combination of logistic modeling, contingency tables, and linear correlations with parameter isolines to determine which parameters had the largest effects on scattering layer boundaries. Although results were inconsistent, we found scattering layer depths to be largely independent of the oxygen content in the water column, and the recent success of *D. gigas* in the Gulf of California is therefore not likely to be attributable to the effects of shoaling oxygen minimum zones on acoustic scattering layers.

11:15

3aAB9. Understanding the relationship between ice, primary producers, and consumers in the Bering Sea. Jennifer L. Miksis-Olds (Appl. Res. Lab, Penn State, PO Box 30, Mailstop 3510D, State College, PA 16804, jlm91@psu.edu) and Stauffer A. Stauffer (Office of Res. and Development, US Environ. Protection Agency, Washington, DC)

Technology has progressed to the level of allowing for investigations of trophic level interactions over time scales of months to years which were previously intractable. A combination of active and passive acoustic technology has been integrated into sub-surface moorings on the Eastern Bering Sea shelf, and seasonal transition measurements were examined to better understand how interannual variability of hydrographic conditions, phytoplankton biomass, and acoustically derived consumer abundance and community structure are related. Ocean conditions were significantly different in 2012 compared to relatively similar conditions in 2009, 2010, and 2011. Differences were largely associated with variations in sea ice extent, thickness, retreat timing, and water column stratification. There was a high degree of variability in the relationships between different classes of consumers and hydrographic condition, and evidence for intra-consumer interactions and trade-offs between different size classes was apparent. Phytoplankton blooms in each year stimulated different components of the consumer population. Acoustic technology now provides the opportunity to explore the ecosystem dynamics in a remote, ice-covered region that was previously limited to ship-board measurements during ice-free periods. The new knowledge we are gaining from remote, long-term observations is resulting in a re-examination of previously proposed ecosystem theories related to the Bering Sea.

11:30

3aAB10. Temporal and spatial patterns of marine soundscape in a coastal shallow water environment. Shane Guan (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13728, Silver Spring, MD 20910, shane.guan@noaa.gov), Tzu-Hao Lin (Inst. of Ecology & Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan), Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, MD), LIen-Siang Chou (Inst. of Ecology & Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan), and John A. Judge (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

Underwater acoustic recordings were made at two coastal shallow water locations, Yunlin (YL) and Waishanding (WS), off Taiwan between June and December 2012. The purpose of the study was to establish soundscape baselines and characterize the acoustic habitat of the critically endangered Eastern Taiwan Strait Chinese white dolphin by investigating: (1) major contributing sources that dominant the soundscape, (2) temporal, spatial, and spectral patterns of the soundscape, and (3) correlations of known sources and their potential effects on dolphins. Results show that choruses from croaker fish (family Sciaenidae) were dominant sound sources in the 1.2–2.4 kHz frequency band for both locations at night, and noises from container ships in the 150–300 Hz frequency band define the relative higher broadband sound levels at YL. In addition, extreme temporal variation in the 150–300 Hz frequency band were observed at WS, which was shows to be linked to the tidal cycle and current velocity. Furthermore, croaker choruses are found to be most intense around the time of high tide at night, but not so around the time of low tide. These results illustrate interrelationships among different biotic, abiotic, and anthropogenic environmental elements that shape the unique fine-scale soundscape in a coastal environment.

Session 3aAO

Acoustical Oceanography, Underwater Acoustics, and Education in Acoustics: Education in Acoustical Oceanography and Underwater Acoustics

Andone C. Lavery, Cochair
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Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Arthur B. Baggeroer, Cochair
Mechanical and Electrical Engineering, Massachusetts Inst. of Technology, Room 5-206, MIT, Cambridge, MA 02139

Chair's Introduction—8:00

Invited Papers

8:05

3aAO1. Ocean acoustics education—A perspective from 1970 to the present. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

A very senior ocean acoustician is attributed with the quote to the effect "one does not start in ocean acoustics, but rather ends up in it." This may well summarize the issues confronting education in ocean acoustics. Acoustics were part of the curriculum in physics departments, whereas now it is spread across many departments. Acoustics and perhaps ocean acoustics are most often found in mechanical or ocean engineering departments, but seldom in physics. Almost all our pioneers from the WWII era were educated in physics and some more recently in engineering departments. Yet, only a few places maintained in depth curricula in ocean acoustics. Most education was done by one on one mentoring. Now the number of students is diminishing, whether because of perception of employment opportunities or the number of available assistantships is uncertain. ONR is the major driver in ocean acoustics for supporting graduate students. The concern about this is hardly new. Twenty plus years ago this was codified as part of the so called "Lackie Report" establishing ocean acoustics as "Navy unique" giving it a priority as a "Navy National Need" (NNR). With fewer students enrolled in ocean acoustics administrators at universities are really balking at sponsoring faculty slots, so there are very significant issues arising for an education in ocean acoustics. Perhaps, reverting to the original model of fundamental training in a related discipline followed by on the job training may be the only option for the future.

8:25

3aAO2. Joint graduate education program: Massachusetts Institute of Technology and Woods Hole Oceanographic Institution. Timothy K. Stanton (Dept. Appl. Ocean. Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, tstanton@whoi.edu)

The 40 + year history of this program will be presented, with a focus on the underwater acoustics and signal processing component. Trends in enrollment will be summarized.

8:35

3aAO3. Graduate studies in underwater acoustics at the University of Washington. Peter H. Dahl, Robert I. Odom, and Jeffrey A. Simmen (Appl. Phys. Lab. and Mech. Eng. Dept., Univ. of Washington, Mech. Eng., 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu)

The University of Washington through its Departments of Mechanical and Electrical Engineering (College of Engineering), Department of Earth and Space Sciences, and School of Oceanography (College of the Environment), and by way of its Applied Physics Laboratory, which links all four of these academic units, offers a diverse graduate education experience in underwater acoustics. A summary is provided of the research infrastructure, primarily made available through Applied Physics Laboratory, which allows for ocean going and arctic field opportunities, and course options offered through the four units that provide the multi-disciplinary background essential for graduate training in the field of underwater acoustics. Students in underwater acoustics can also mingle in or extend their interests into medical acoustics research. Degrees granted include both the M.S and Ph.D.

8:45

3aAO4. Acoustical Oceanography and Underwater Acoustics; their role in the Pennsylvania State University Graduate Program in Acoustics. David Bradley (Penn State Univ., PO Box 30, State College, PA 16870, dlb25@psu.edu) and Victor Sparrow (Penn State Univ., University Park, PA)

The Pennsylvania State University Graduate Program in Acoustics has a long and successful history in Acoustics Education. A brief history together with the current status of the program will be discussed. An important aspect of the program has been the strong role of the Applied Research Laboratory, both in support of the program as well as for the graduate students enrolled. Presentation includes details of course content, variability to fit student career goals and program structure, including resident and distance education opportunities. The future of the program at Penn State will also be addressed.

8:55

3aAO5. Ocean acoustics at the University of Victoria. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

This paper describes the academic program in Ocean Acoustics and Acoustical Oceanography at the University of Victoria in Canada. The program was established when a Research Chair in Ocean Acoustics consisting of two faculty members was funded in 1995 by the Canadian Natural Sciences and Engineering Research Council (NSERC). The Research Chair graduate program offered two courses in Ocean Acoustics, and courses in Time Series Analysis and Inverse Methods. Funding for students was obtained entirely through partnership research programs with Canadian marine industry, the Department of National Defence in Canada and the Office of Naval Research. The program was successful in graduating around 30 M.Sc. and Ph.D. students to date, about half of whom were Canadians. Notably, all the students obtained positions in marine industry, government, or academia after their degrees. The undergraduate program consisted of one course in Acoustical Oceanography at the senior level (3rd year) that was designed to appeal to students in physics, biology, and geology. The course attracted about 30 students each time, primarily from biology. The paper concludes with perspectives on difficulties in operating an academic program with a low critical mass of faculty and in isolation from colleagues in the research field.

9:05

3aAO6. Ocean acoustics away from the ocean. David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133, drd@umich.edu)

Acoustics represents a small portion of the overall educational effort in engineering and science, and ocean acoustics is one of many topic areas in the overall realm of acoustics. Thus, maintaining teaching and research efforts involving ocean acoustics is challenging but not impossible, even at a university that is more than 500 miles from the ocean. This presentation describes the author's two decades of experience in ocean acoustics education and research. Success is possible by first attracting students to acoustics, and then helping them wade into a research topic in ocean acoustics that overlaps with their curiosity, ambition, or both. The first step occurs almost naturally since college students' experience with their ears and voice provides intuition and motivation that allows them to readily grasp acoustic concepts and to persevere through mathematical courses. The second step is typically no more challenging since ocean acoustics is a leading and fascinating research area that provides stable careers. Plus, there are even some advantages to studying ocean acoustics away from the ocean. For example, matched-field processing, a common ocean acoustic remote sensing technique, appears almost magical to manufacturing or automotive engineers when applied to assembly line and safety problems involving airborne sound.

9:15

3aAO7. Office of Naval Research special research awards in ocean acoustics. Robert H. Headrick (Code 32, Office of Naval Res., 875 North Randolph St., Arlington, VA 22203, bob.headrick@navy.mil)

The Ocean Acoustics Team of the Office of Naval Research manages the Special Research Awards that support graduate traineeship, postdoctoral fellowship, and entry-level faculty awards in ocean acoustics. The graduate traineeship awards provide for study and research leading to a doctoral degree and are given to individuals who have demonstrated a special aptitude and desire for advanced training in ocean acoustics or the related disciplines of undersea signal processing, marine structural acoustics and transducer materials science. The postdoctoral fellowship and entry-level faculty awards are similarly targeted. These programs were started as a component of the National Naval Responsibility in Ocean Acoustics to help ensure a stable pipeline of talented individuals would be available to support the needs of the Navy in the future. They represent only a fraction of the students, postdocs, and early faculty researchers that are actively involved the basic research supported by the Ocean Acoustics Program. A better understanding of the true size of the pipeline and the capacity of the broader acoustics related Research and Development community to absorb the output is needed to maintain a balance in priorities for the overall Ocean Acoustics Program.

9:25

3aAO8. Underwater acoustics education at the University of Texas at Austin. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu), Mark F. Hamilton (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Clark S. Penrod, Frederick M. Pestorius (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The University of Texas at Austin has supported education and research in acoustics since the 1930s. The Cockrell School of Engineering currently offers a wide range of graduate courses and two undergraduate courses in acoustics, not counting the many courses in hearing, speech, seismology, and other areas of acoustics at the university. An important adjunct to the academic program in acoustics has been the Applied Research Laboratories (ARL). Spun off in 1945 from the WW II Harvard Underwater Sound Laboratory (1941–1949) and founded as the Defense Research Laboratory, ARL is one of five University Affiliated Research Centers formally recognized by the US Navy for their prominence in underwater acoustics research and development. ARL is an integral part of UT Austin, and this

symbiotic combination of graduate and undergraduate courses, and laboratory and field work, provides one of the leading underwater acoustics education programs in the nation. In this talk, the underwater acoustics education program will be described with special emphasis on the underwater acoustics course and its place in the larger acoustics program. Statistics on education, funding, and placement of graduate students in the program will also be presented.

9:35

3aAO9. Acoustical Oceanography and Underwater Acoustics Graduate Programs at the Scripps Institution of Oceanography of the University of California, San Diego. William A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Marine Physical Lab., La Jolla, CA 92093-0238, wkuperman@ucsd.edu)

The Scripps Institution of Oceanography (SIO) of the University of California, San Diego (UCSD), has graduate programs in all areas of acoustics that intersect oceanography. These programs are associated mostly with internal SIO divisions that include the Marine Physical Laboratory, Physical Oceanography, Geophysics, and Biological Oceanography as well as SIO opportunities for other UCSD graduate students in the science and engineering departments. Course work includes basic wave physics, graduate mathematics, acoustics and signal processing, oceanography and biology, digital signal processing, and geophysics/seismology. Much of the emphasis at SIO includes at-sea experience. Recent examples of thesis research has been in marine mammal acoustics, ocean tomography and seismic/acoustic inversion methodology, acoustical signal processing, ocean ambient noise inversion, ocean/acoustic exploration, and acoustic sensing of the air-sea interaction. An overview of the SIO/UCSD graduate program is presented.

9:45

3aAO10. Underwater acoustics education at Portland State University. Martin Siderius and Lisa M. Zurk (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, siderius@pdx.edu)

The Northwest Electromagnetics and Acoustics Research Laboratory (NEAR-Lab) is in the Electrical and Computer Engineering Department at Portland State University (PSU) in Portland, Oregon. The NEAR-Lab was founded in 2005 and is co-directed by Lisa M. Zurk and Martin Siderius. A primary interest is underwater acoustics, and students at undergraduate and graduate levels (occasionally also high school students) regularly participate in research. This is synergistic with underwater acoustics education at PSU, which includes a course curriculum that provides opportunities for theoretical and experimental research and multiple course offerings at both the undergraduate and graduate level. The research generally involves modeling and analysis of acoustic propagation and scattering, acoustic signal processing, algorithm development, environmental acoustics, and bioacoustics. The lab maintains a suite of equipment for experimentation including hydrophone arrays, sound projectors, a Webb Slocum glider, an electronics lab, and an acoustic tank. Large-scale experiments that include student participation have been routinely conducted by successful collaborations such as with the APL-University of Washington, NATO Centre for Maritime Research and Experimentation, and the University of Hawaii. In this talk, the state of the PSU underwater acoustics program will be described along with the courses offered, research activities, experimental program, collaborations, and student success.

9:55

3aAO11. Underwater acoustics education in Harbin Engineering University. Desen Yang, Xiukun Li, and Yang Li (Acoust. Sci. and Technol. Lab., Harbin Eng. Univ., Harbin, Heilongjiang Province, China, dsyang@hrbeu.edu.cn)

College of Underwater Acoustic Engineering in Harbin Engineering University is the earliest institution engaging in underwater acoustics education in Chinese universities, which has complete types of high education training levels and subject directions. There are 124 teachers in the college engaging in underwater acoustics research, of which there are 30 professors and 36 associate professors. The developments of underwater acoustic transducer technology, underwater positioning and navigation technology, underwater target detecting technology, underwater acoustic communication technique, multi-beam echo sounding technique, and high resolution image sonar technique new theory and technology of underwater acoustic reach the leading level in China. Every year, the college attracts more than 200 excellent students whose entrance examination marks is 80 points higher than the key fraction stroke. There are three education program levels in this specialty (undergraduate-level, graduate-level, and Ph.D.-level), and students may study underwater acoustics within any of our three programs, besides which, the college has special education programs for foreign students. Graduate employments are underwater acoustic institution, electronic institution, communication company, and IT enterprise. In this paper, descriptions of underwater acoustics education programs, curriculum systems, and teaching contents of acoustics courses will be introduced.

10:05–10:20 Break

Contributed Papers

10:20

3aAO12. Graduate education in underwater acoustics, transduction, and signal processing at UMass Dartmouth. David A. Brown (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net), John Buck, Karen Payton, and Paul Gendron (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

The University of Massachusetts Dartmouth established a Ph.D. degree in Electrical Engineering with a specialization in Marine Acoustics in 1996, building on the strength of the existing M.S. program. Current enrollment in these programs include 26 M.S. students and 16 Ph.D. students. The program offers courses and research opportunities in the area of underwater acoustics, transduction, and signal processing. Courses include the Fundamentals of Acoustics, Random Signals, Underwater Acoustics, Introduction to Transducers, Electroacoustic Transduction, Digital Signal Processing, Detection Theory, and Estimation Theory. The university's indoor underwater acoustic test and calibration facility is one of the largest in academia and supports undergraduate and graduate thesis and sponsored research. The university also owns three Iver-2 fully autonomous underwater vehicles. The graduate program capitalizes on collaborations with many marine technology companies resident at the university's Advanced Technology and

Manufacturing Center (ATMC) and the nearby Naval Undersea Warfare Center in Newport, RI. The presentation will highlight recent theses and dissertations, course offerings, and industry and government collaborations that support underwater acoustics research.

10:30

3aAO13. Ocean acoustics at the University of Rhode Island. 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)

The undergraduate and graduate program in Ocean Engineering at the University of Rhode Island is one of the oldest such programs in the United States. This program offers Bachelors, Masters (thesis and non-thesis options), and Ph.D. degrees. At the undergraduate level, students are exposed to ocean acoustics through a number of required and elective courses, laboratory and field work, and capstone projects. Examples of student projects will be presented. At the graduate level, students can specialize in several areas including geoacoustic inversion, propagation modeling, marine mammal acoustics, ocean acoustic instrumentation, transducers, etc. A historical review of the evolution of ocean acoustics education in the department will be presented. This will include examples of some of the research carried out by different faculty and students over the years, enrollment trends, collaborations, external funding, etc. Many graduates from the program hold faculty positions at a number of universities in the US and abroad. In addition, graduates from the ocean acoustics program at URI are key staff at many companies and organizations. A number of companies have spun off the program in the areas of forward-looking sonar, sub-bottom profiling, and other applications. The opportunities and challenges facing the program will be summarized.

10:40

3aAO14. An underwater acoustics program far from the ocean: The Georgia Tech case. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

The underwater acoustics education program at the Georgia Institute of Technology (Georgia Tech) is run by members of the Acoustics and Dynamics research area group from the School of Mechanical Engineering. We will briefly review the scope of this program in terms of education and research activities as well as discuss current challenges related to the future of underwater acoustics education.

10:50

3aAO15. Graduate education in ocean acoustics at Rensselaer Polytechnic Institute. William L. Siegmann (Dept. of Mathematical Sci., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180-3590, siegmw@rpi.edu)

Doctoral and master's students in Rensselaer's Department of Mathematical Sciences have had opportunities for research in Ocean Acoustics since 1957. Since then only one or two faculty members at any time were directly involved with OA education. Consequently, collaboration with colleagues at other centers of OA research has been essential. The history will be briefly reviewed, focusing on the education of a small group of OA doctoral students in an environment with relatively limited institutional resources. Graduate education in OA at RPI has persisted because of sustained support by the Office of Naval Research.

11:00–11:45 Panel Discussion

11:45

3aAO16. Summary of panel discussion on education in Acoustical Oceanography and Underwater Acoustics. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

Following the presentations by the speakers in session, a panel discussion will offer the platform for those in the audience, particularly those from Institutions and Universities that did not formally participate in the session but have active education programs in Acoustical Oceanography and/or Underwater Acoustics, to ask relevant questions and contribute to the assessment of the national health of education in the fields of Acoustical Oceanography and Underwater Acoustics. A summary of the key points presented in the special sessions and panel discussion is provided.

Session 3aBA

Biomedical Acoustics: Kidney Stone Lithotripsy

Tim Colonius, Cochair
Mechanical Engineering, Caltech, 1200 E. California Blvd., Pasadena, CA 91125

Wayne Kreider, Cochair
CIMU, Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Contributed Papers

8:00

3aBA1. Comparable clinical outcomes with two lithotripters having substantially different acoustic characteristics. James E. Lingeman, Naeem Bhojani (Urology, Indiana Univ. School of Medicine, 1801 N. Senate Blvd., Indianapolis, IN 46202, jlingeman@iuhealth.org), James C. Williams, Andrew P. Evan, and James A. McAteer (Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN)

A consecutive case study was conducted to assess the clinical performance of the Lithogold, an electrohydraulic lithotripter having a relatively low P+ and broad focal width (FW) (~20 MPa, ~20 mm), and the electromagnetic Storz-SLX having higher P+ and narrower FW (~50 MPa, 3–4 mm). Treatment was at 60 SW/min with follow-up at ~2 weeks. Stone free rate (SFR) was defined as no residual fragments remaining after single session SWL. SFR was similar for the two lithotripters (Lithogold 29/76 = 38.2%; SLX 69/142 = 48.6% p = 0.15), with no difference in outcome for renal stones (Lithogold 20/45 = 44.4%; SLX 33/66 = 50%, p = 0.70) or stones in the ureter (Lithogold 9/31 = 29%; SLX 36/76 = 47.4%, p = 0.08). Stone size did not differ between the two lithotripters for patients who were not stone free (9.1±3.7 mm for Lithogold vs. 8.5±3.5 mm for SLX, P = 0.42), but the stone-free patients in the Lithogold group had larger stones on average than the stone-free patients treated with the SLX (7.6±2.5 mm vs. 6.2±3.2 mm, P = 0.005). The percentage of stones that did not break was similar (Lithogold 10/76 = 13.2%; SLX 23/142 = 16.2%). These data present a realistic picture of clinical outcomes using modern lithotripters, and although the acoustic characteristics of the Lithogold and SLX differ considerably, outcomes were similar. [NIH-DK43881.]

8:15

3aBA2. Characterization of an electromagnetic lithotripter using transient acoustic holography. Oleg A. Sapozhnikov, Sergey A. Tsysar (Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, oa.sapozhnikov@gmail.com), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Guangyan Li (Dept. of Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN), Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Michael R. Bailey (Dept. of Urology, Univ. of Washington Medical Ctr., Seattle, WA)

Shock wave lithotripters radiate high intensity pulses that are focused on a kidney stone. High pressure, short rise time, and path-dependent nonlinearity make characterization in water and extrapolation to tissue difficult. Here acoustic holography is applied for the first time to characterize a lithotripter. Acoustic holography is a method to determine the distribution of acoustic pressure on the surface of the source (source hologram). The electromagnetic lithotripter characterized in this effort is a commercial model (Dornier Compact S, Dornier MedTech GmbH, Wessling, Germany) with 6.5 mm focal width. A broadband hydrophone (HGL-0200, sensitive diameter 200 μm, Onda Corp., Sunnyvale, CA) was used to sequentially measure the field over a set of points in a plane in front of the source. Following the previously developed transient holography approach, the recorded pressure field was numerically back-propagated to the source surface and then used for nonlinear forward propagation to predict waveforms in different points in the focal region. Pressure signals predicted from the source hologram coincide well with the waveforms measured by a fiber optic hydrophone. Moreover, the method provides an accurate boundary condition from which the field in tissue can be simulated. [Work supported by RSF 14-15-00665 and NIH R21EB016118, R01EB007643, and DK043881.]

8:30

3aBA3. Multiscale model of comminution in shock wave lithotripsy. Sorin M. Mitran (Mathematics, Univ. of North Carolina, CB 3250, Chapel Hill, NC 27599-3250, mitran@amath.unc.edu), Georgy Sankin, Ying Zhang, and Pei Zhong (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

A previously introduced model for stone comminution in shock wave lithotripsy is extended to include damage produced by cavitation. At the macroscopic, continuum level a 3D elasticity model with time-varying material constants capturing localized damage provides the overall stress field within kidney stone simulants. Regions of high stress are identified and a mesoscopic crack propagation is used to dynamically update localized damage. The crack propagation model in turn is linked with a microscopic grain dynamics model. Continuum stresses and surface pitting is provided by a multiscale cavitation model (see related talk). The overall procedure is capable of tracking stone fragments and surface cavitation of the fragments through several levels of breakdown. Computed stone fragment distributions are compared to experimental results. [Work supported by NIH through 5R37DK052985-18.]

8:45

3aBA4. Exploring the limits of treatment used to invoke protection from extracorporeal shock wave lithotripsy induced injury. Bret A. Connors, Andrew P. Evan, Rajash K. Handa, Philip M. Blomgren, Cynthia D. Johnson, James A. McAteer (Anatomy and Cell Biology, IU School of Medicine, Medical Sci. Bldg., Rm. 5055, 635 Barnhill Dr., Indianapolis, IN 46202, bconnors@iupui.edu), and James E. Lingeman (Urology, IU School of Medicine, Indianapolis, IN)

Previous studies with our juvenile pig model have shown that a clinical dose of 2000 shock waves (SWs) (Dornier HM-3, 24 kV, 120 SWs/min) produces a lesion ~3–5% of the functional renal volume (FRV) of the SW-treated kidney. This injury was significantly reduced (to ~0.4% FRV) when a priming dose of 500 low-energy SWs immediately preceded this clinical dose, but not when using a priming dose of 100 SWs [BJU Int. 110, E1041 (2012)]. The present study examined whether using only 300 priming dose SWs would initiate protection against injury. METHODS: Juvenile pigs were treated with 300 SW's (12 kV) delivered to a lower pole calyx using a HM-3 lithotripter. After a pause of 10 s, 2000 SWs (24 kV) were delivered

3a WED. AM

to that same kidney. The kidneys were then perfusion-fixed and processed to quantitate the size of the parenchymal lesion. RESULTS: Pigs (n = 9) treated using a protocol with 300 low-energy priming dose SWs had a lesion measuring 0.84±0.43% FRV (mean ± SE). This lesion was smaller than that seen with a clinical dose of 2000 SWs at 24 kV. CONCLUSIONS: A treatment protocol including 300 low-energy priming dose SWs can provide protection from injury during shock wave lithotripsy. [Research supported by NIH grant P01 DK43881.]

9:00

3aBA5. Shockwave lithotripsy with renoprotective pause is associated with vasoconstriction in humans. Franklin Lee, Ryan Hsi, Jonathan D. Harper (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Barbrina Dunmire, Michael Bailey (Cntr.Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), Ziyue Liu (Dept. of Biostatistics, Indiana Univ. School of Medicine, Indianapolis, Washington), and Mathew D. Sorensen (Dept. of Urology, Dept. of Veteran Affairs Medical Ctr., Seattle, WA)

A pause early in shock wave lithotripsy (SWL) increased vasoconstriction as measured by resistive index (RI) during treatment and mitigated renal injury in an animal model. The purpose of our study was to investigate whether RI rose during SWL in humans. Prospectively recruited patients underwent SWL of renal stones with a Dornier Compact S lithotripter. The renal protective protocol consisted of treatment at 1 Hz and slow power ramping for the initial 250 shocks followed by a 2 min pause. RI was measured using ultrasound prior to treatment, after 250 shocks, after 750 shocks, after 1500 shocks, and after SWL. A linear mixed-effects model was used to compare RI at the different time points and to account for additional covariates in fifteen patients. RI was significantly higher than baseline for all time points 250 shocks and after. Age, gender, body mass index, and treatment side were not significantly associated with RI. Monitoring for a rise in RI during SWL is possible and may provide real-time feedback as to when the kidney is protected. [Work supported by NIH DK043881, NSBRI through NASA NCC 9-58, and resources from the VA Puget Sound Health Care System.]

9:15

3aBA6. Renal shock wave lithotripsy may be a risk factor for early-onset hypertension in metabolic syndrome: A pilot study in a porcine model. Rajash Handa (Anatomy & Cell Biology, Indiana Univ. School of Medicine, 635 Barnhill Dr., MS 5035, Indianapolis, IN 46202-5120, rhanda@iupui.edu), Ziyue Liu (Biostatistics, Indiana Univ. School of Medicine, Indianapolis, IN), Bret Connors, Cynthia Johnson, Andrew Evan (Anatomy & Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN), James Lingeman (Kidney Stone Inst., Indiana Univ. Health Methodist Hospital, Indianapolis, IN), David Basile, and Johnathan Tune (Cellular & Integrative Physiol., Indiana Univ. School of Medicine, Indianapolis, IN)

A pilot study was conducted to assess whether extracorporeal shock wave lithotripsy (SWL) treatment of the kidney influences the onset and severity of metabolic syndrome (MetS)—a cluster of conditions that includes central obesity, insulin resistance, impaired glucose tolerance, dyslipidemia, and hypertension. Methods: Three-month-old juvenile female Ossabaw miniature pigs were treated with either SWL (2000 SWs, 24 kV, 120 SWs/min using the HM3 lithotripter; n = 2) or sham-SWL (no SWs; n = 2). SWs were targeted to the upper pole of the left kidney so as to model treatment that would also expose the pancreas—an organ involved in blood glucose homeostasis—to SWs. The pigs were then instrumented for direct measurement of arterial blood pressure via implanted radiotelemetry devices, and later fed a hypercaloric atherogenic diet for ~7 months to induce MetS. The development of MetS was assessed from intravenous glucose tolerance tests. Results: The progression and severity of MetS were similar in the sham-treated and SWL-treated groups. The only exception was arterial blood pressure, which remained relatively constant in the sham-treated pigs and rose toward hypertensive levels in SW-treated pigs. Conclusions. These preliminary results suggest that renal SWL appears to be a risk factor for early-onset hypertension in MetS.

9:30–9:45 Break

9:45

3aBA7. Modeling vascular injury due to shock-induced bubble collapse in lithotripsy. Vedran Coralic and Tim Colonius (Mech. Eng., Caltech, 1200 E. California Blvd., Pasadena, CA 91125, colonius@caltech.edu)

Shock-induced collapse (SIC) of preexisting bubbles is investigated as a potential mechanism for vascular injury in shockwave lithotripsy (SWL). Preexisting bubbles exist under normal physiological conditions and grow larger and more numerous with ongoing treatment. We compute the three-dimensional SIC of a bubble using the multi-component Euler equations, and determine the resulting three-dimensional finite-strain deformation field in the material surrounding the collapsing bubble. We propose a criterion for vessel rupture and estimate the minimum bubble size, across clinical SWL pressures, which could result in rupture of microvasculature. Post-processing of the results and comparison to viscoelastic models for spherical bubble dynamics demonstrate that our results are insensitive to a wide range of estimated viscoelastic tissue properties during the collapse phase. During the jetting phase, however, viscoelastic effects are non-negligible. The minimum bubble size required to rupture a vessel is then estimated by adapting a previous model for the jet's penetration depth as a function of tissue viscosity.

10:00

3aBA8. Multiscale model of cavitation bubble formation and breakdown. Isaac Nault, Sorin M. Mitran (Mathematics, Univ. of North Carolina, CB3250, Chapel Hill, NC, naulti@live.unc.edu), Georgy Sankin, and Pei Zhong (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

Cavitation damage is responsible for initial pitting of kidney stone surfaces, damage that is thought to play an important role in shock wave lithotripsy. We introduce a multiscale model of the formation of cavitation bubbles in water, and subsequent breakdown. At a macroscopic, continuum scale cavitation is modeled by the 3D Euler equations with a Tait equation of state. Adaptive mesh refinement is used to provide increased resolution at the liquid/vapor boundary. Cells with both liquid and vapor phases are flagged by the continuum solver for mesoscale, kinetic modeling by a lattice Boltzmann description capable of capturing non-equilibrium behavior (e.g., phase change, energetic jet impingement). Isolated and interacting two-bubble configurations are studied. Computational simulation results are compared with high-speed experimental imaging of individual bubble dynamics and bubble–bubble interaction. The model is used to build a statistical description of multiple-bubble interaction, with input from cavitation cloud imaging. [Work supported by NIH through 5R37DK052985-18.]

10:15

3aBA9. Preliminary results of the feasibility to reposition kidney stones with ultrasound in humans. Jonathan D. Harper, Franklin Lee, Susan Ross, Hunter Wessells (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Bryan W. Cunitz, Barbrina Dunmire, Michael Bailey (Ctr.Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), Jeff Thiel (Dept. of Radiology, Univ. of Washington School of Medicine, Seattle), Michael Coburn (Dept. of Urology, Baylor College of Medicine, Houston, TX), James E. Lingeman (Dept. of Urology, Indiana Univ. School of Medicine, Indianapolis, IN), and Mathew Sorensen (Dept. of Urology, Dept. of Veteran Affairs Medical Ctr., Seattle)

Preliminary investigational use of ultrasound to reposition human kidney stones is reported. The three study arms include: *de novo* stones, post-lithotripsy fragments, and large stones within the preoperative setting. A pain questionnaire is completed immediately prior to and following propulsion. A maximum of 40 push attempts are administered. Movement is classified as no motion, movement with rollback or jiggle, or movement to a new location. Seven subjects have been enrolled and undergone ultrasonic propulsion to date. Stones were identified, targeted, and moved in all subjects. Subjects who did not have significant movement were in the *de novo* arm. None of the subjects reported pain associated with the treatment. One subject in the post-lithotripsy arm passed two small stones immediately following treatment corresponding to the two stones displaced from the interpolar region. Three post-lithotripsy subjects reported passage of multiple small fragments within two weeks of treatment. In four subjects, ultrasonic

propulsion identified a collection of stones previously characterized as a single stone on KUB and ultrasound. There have been no treatment related adverse events reported with mean follow-up of 3 months. [Trial supported by NSBRI through NASA NCC 9-58. Development supported by NIH DK043881and DK092197.]

10:30

3aBA10. Nonlinear saturation effects in ultrasound fields of diagnostic-type transducers used for kidney stone propulsion. Maria M. Karzova (Phys. Faculty, Dept. of Acoust., M.V. Lomonosov Moscow State Univ., Leninskie Gory 1/2, Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Bryan W. Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Petr V. Yuldashev (Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Vera A. Khokhlova, Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Michael R. Bailey (Dept. of Urology, Univ. of Washington Medical Ctr., Seattle, WA)

A novel therapeutic application of ultrasound for repositioning kidney stones is being developed. The method uses acoustic radiation force to expel mm-sized stones or to dislodge even larger obstructing stones. A standard diagnostic 2.3 MHz C5-2 array probe has been used to generate pushing acoustic pulses. The probe comprises 128 elements equally spaced at the 55 mm long convex cylindrical surface with 41.2 mm radius of curvature. The efficacy of the treatment can be increased by using higher transducer output to provide stronger pushing force; however, nonlinear acoustic saturation effect can be a limiting factor. In this work, nonlinear propagation effects were analyzed for the C5-2 transducer using a combined measurement and modeling approach. Simulations were based on the 3D Westervelt equation; the boundary condition was set to match low power measurements. Focal waveforms simulated for several output power levels were compared with the fiber-optic hydrophone measurements and were found in good agreement. It was shown that saturation effects do limit the acoustic pressure in the focal region of the transducer. This work has application to standard diagnostic probes and imaging. [Work supported by RSF 14-12-00974, NIH EB007643, DK43881 and DK092197, and NSBRI through NASA NCC 9-58.]

10:45

3aBA11. Evaluating kidney stone size in children using the posterior acoustic shadow. Franklin C. Lee, Jonathan D. Harper, Thomas S. Lendvay (Urology, Univ. of Washington, Seattle, WA), Ziyue Liu (Biostatistics, Indiana Univ. School of Medicine , Indianapolis, IN), Barbrina Dunmire (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105, mrbean@uw.edu), Manjiri Dighe (Radiology, Univ. of Washington, Seattle, WA), Michael R. Bailey (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Mathew D. Sorensen (Urology, Dept. of Veteran Affairs Medical Ctr., Seattle, WA)

Ultrasound, not x-ray, is preferred for imaging kidney stones in children; however, stone size determination is less accurate with ultrasound. *In vitro* we found stone sizing was improved by measuring the width of the acoustic shadow behind the stone. We sought to determine the prevalence and accuracy of the acoustic shadow in pediatric patients. A retrospective analysis was performed of all initial stone events at a children's hospital over the last 10 years. Included subjects had a computed tomography (CT) scan and renal ultrasound within 3 months of each other. The width of the stone and acoustic shadow were measured on ultrasound and compared to the stone size as determined by CT. Thirty-seven patients with 49 kidney stones were included. An acoustic shadow was seen in 85% of stones evaluated. Stone width resulted in an average overestimation of 1.2 ± 2.2 mm while shadow width resulted in an underestimation of 0.5 ± 1.7 mm ($p < 0.001$). A posterior acoustic shadow was seen in the majority of stones and was a more accurate measure of stone size. This would provide valuable information for stone management. [Work supported by NIH DK43881 and DK092197, and NSBRI through NASA NCC 9-58.]

11:00

3aBA12. Development and testing of an image-guided prototype system for the comminution of kidney stones using burst wave lithotripsy. Bryan Cunitz (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bwc@apl.washington.edu), Adam Maxwell (Dept. of Urology, Univ. of Washington Medical Ctr., Seattle, WA), Wayne Kreider, Oleg Sapozhnikov (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Franklin Lee, Jonathan Harper, Matthew Sorenson (Dept. of Urology, Univ. of Washington Medical Ctr., Seattle, WA), and Michael Bailey (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Burst wave lithotripsy is a novel technology that uses focused, sinusoidal bursts of ultrasound to fragment kidney stones. Prior research laid the groundwork to design an extracorporeal, image-guided probe for *in-vivo* testing and potentially human clinical testing. Toward this end, a 12-element 330 kHz array transducer was designed and built. The probe frequency, geometry, and shape were designed to break stones up to 1 cm in diameter into fragments <2mm. A custom amplifier capable of generating output bursts up to 3 kV was built to drive the array. To facilitate image guidance, the transducer array was designed with a central hole to accommodate co-axial attachment of an HDI P4-2 probe. Custom B-mode and Doppler imaging sequences were developed and synchronized on a Verasonics ultrasound engine to enable real-time stone targeting and cavitation detection, Preliminary data suggest that natural stones will exhibit Doppler "twinkling" artifact in the BWL focus and that the Doppler power increases as the stone begins to fragment. This feedback allows accurate stone targeting while both types of imaging sequences can also detect cavitation in bulk tissue that may lead to injury. [Work supported by NIH grants DK043881, EB007643, EB016118, T32 DK007779, and NSBRI through NASA NCC 9-58.]

11:15

3aBA13. Removal of residual bubble nuclei to enhance histotripsy kidney stone erosion at high rate. Alexander P. Duryea (Biomedical Eng., Univ. of Michigan, 2131 Gerstacker Bldg., 2200 Bonisteel Blvd., Ann Arbor, MI 48109, duryalex@umich.edu), William W. Roberts (Urology, Univ. of Michigan, Ann Arbor, MI), Charles A. Cain, and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Previous work has shown that histotripsy can effectively erode model kidney stones to tiny, sub-millimeter debris via a cavitational bubble cloud localized on the stone surface. Similar to shock wave lithotripsy, histotripsy stone treatment displays a rate-dependent efficacy, with pulses applied at low repetition frequency producing more efficient erosion compared to those applied at high repetition frequency. This is attributed to microscopic residual cavitation bubble nuclei that can persist for hundreds of milliseconds following bubble cloud collapse. To mitigate this effect, we have developed low amplitude (MI<1) acoustic pulses to actively remove residual nuclei from the field. These bubble removal pulses utilize the Bjerknes forces to stimulate the aggregation and subsequent coalescence of remnant nuclei, consolidating the population from a very large number to a countably small number of remnant bubbles within several milliseconds. Incorporation of this bubble removal scheme in histotripsy model stone treatments performed at high rate (100 pulses/second) produced drastic improvement in treatment efficiency, with an average erosion rate increase of 12-fold in comparison to treatment without bubble removal. High speed imaging indicates that the influence of remnant nuclei on the location of bubble cloud collapse is the dominant contributor to this disparity in treatment efficacy.

Session 3aEA

Engineering Acoustics and Structural Acoustics and Vibration: Mechanics of Continuous Media

Andrew J. Hull, Cochair
Naval Undersea Warfare Center, 1176 Howell St, Newport, RI 02841

J. Gregory McDaniel, Cochair
Mechanical Engineering, Boston Univ., 110 Cummington St., Boston, MA 02215

Invited Papers

8:00

3aEA1. Fundamental studies of zero Poisson ratio metamaterials. Elizabeth A. Magliula (Div. Newport, Naval Undersea Warfare Ctr., 1176 Howell St., Bldg. 1302, Newport, RI 02841, elizabeth.magliula@navy.mil), J. Gregory McDaniel, and Andrew Wixom (Mech. Eng. Dept., Boston Univ., Boston, MA)

As material fabrication advances, new materials with special properties will be possible to accommodate new design boundaries. An emerging and promising field of investigation is to study the basic phenomena of materials with a negative Poisson ratio (NPR). This work seeks to develop zero Poisson ratio (ZPR) metamaterials for use in reducing acoustic radiation from compressional waves. Such a material would neither contract or expand laterally when compressed or stretched, and therefore not radiate sound. Previous work has provided procedures for creating NPR copper foam through transformation of the foam cell structure from a convex polyhedral shape to a concave "re-entrant" shape. A ZPR composite will be developed and analyzed in an effort to achieve desired wave propagation characteristics. Dynamic investigations have been conducted using ABAQUS, in which a ZPR is placed under load to observe displacement behavior. Inspection of the results at 1 kHz and 5 kHz show that the top and bottom surfaces experience much less displacement compared to respective conventional reference layer build-ups. However, at 11 kHz small lateral displacements were experienced at the outer surfaces. Results indicate that the net zero Poisson effect was successfully achieved at frequencies where half the wavelength is greater than the thickness.

8:20

3aEA2. Scattering by targets buried in elastic sediment. Angie Sarkissian, Saikat Dey, Brian H. Houston (Code 7130, Naval Res. Lab., Code 7132, 4555 Overlook Ave. S.W., Washington, DC 20375, angie.sarkissian@nrl.navy.mil), and Joseph A. Bucaro (Excet, Inc., Springfield, VA)

Scattering results are presented for targets of various shapes buried in elastic sediment with a plane wave incident from air above. The STARS3D finite element program recently extended to layered, elastic sediments is used to compute the displacement field just below the interface. Evidence of the presence of Rayleigh waves is observed in the elastic sediment and an algorithm based on the Rayleigh waves subtracts the contribution of the Rayleigh waves to simplify the resultant scattering pattern. Results are presented for scatterers buried in uniform elastic media as well as layered media. [This work was supported by ONR.]

8:40

3aEA3. Response shaping and scale transition in dynamic systems with arrays of attachments. Joseph F. Vignola, Aldo A. Glean (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), John Sterling (Carderock Div., Naval Surface Warfare Ctr., West Bethesda, MD), and John A. Judge (Mech. Eng., The Catholic Univ. of America, Washington, DC)

Arrays of elastic attachments can be design to act as energy sinks in dynamic systems. This presentation describes design strategies for drawing off mechanical energy to achieve specific objectives such as mode suppression and response tailoring in both extended and discrete systems. The design parameters are established using numerical simulations for both propagating and standing compressional waves in a one-dimensional system. The attachments were chosen to be cantilevers so that higher modes would have limited interaction with the higher modes of the primary structure. The two cases considered here are concentrated groups of cantilevers and spatial distributions of similar cantilevers. Relationships between the number and placement of the attachments and their masses and frequency distributions are of particular interest, along with the energy density distribution between the primary structure and the attachments. The simulations are also used to show how fabrication error degrades performance and how energy scale transition can be managed to maintain linear behavior.

9:00

3aEA4. Accelerated general method for computing noise effects in arrays. Heather Reed, Jeffrey Cipolla, Mahesh Bailakanavar, and Patrick Murray (Weidlinger Assoc., 40 Wall St 18th Fl., New York, NY 10005, heather.reed@wai.com)

Noise in an acoustic array can be defined as any unwanted signal, and understanding how noise interacts with a structural system is paramount for optimal design. For example, in an underwater vehicle we may want to understand how structural vibrations radiate through a surrounding fluid; or an engineer may want to evaluate the level of sound inside a car resulting from the turbulent boundary layer (TBL) induced by a moving vehicle. This talk will discuss a means of modeling noise at a point of interest (e.g., at a sensor location) stemming from a known source by utilizing a power transfer function between the source and the point of interest, a generalization of the work presented in [1]. The power transfer function can be readily computed from the acoustic response to an incident wave field, requiring virtually no additional computation. The acoustic solution may be determined via analytic frequency domain approaches or through a finite element analysis, enabling the noise solution to be a fast post processing exercise. This method is demonstrated by modeling the effects of a TBL pressure and noise induced by structural vibrations on a sensor array embedded in an elastic, multi-layer solid. Additionally, uncertainties in the noise model can be easily quantified through Monte Carlo techniques due to the fast evaluation of the noise spectrum. Ko, S.H. and Schloemer, H.H. "Flow noise reduction techniques for a planar array of hydrophones," J. Acoust. Soc. Am. 92, 3409 (1992).

9:20

3aEA5. Response of distributed fiber optic sensor cables to spherical wave incidence. Jeffrey Boisvert (NAVSEA Div. Newport, 1176 Howell St., Newport, RI 02841, cboisvertj@cox.net)

A generalized multi-layered infinite-length fiber optic cable is modeled using the exact theory of three-dimensional elasticity in cylindrical coordinates. A cable is typically composed of a fiber optic (glass) core surrounded by various layered materials such as plastics, metals, and elastomers. The cable is excited by an acoustic spherical wave radiated by a monopole source at an arbitrary location in the acoustic field. For a given source location and frequency, the radial and axial strains within the cable are integrated over a desired sensor zone length to determine the optical phase sensitivity using an equation that relates the strain distribution in an optical fiber to changes in the phase of an optical signal. Directivity results for the cable in a free-field water environment are presented at several frequencies for various monopole source locations. Some comparisons of the sensor directional response resulting from nearfield (spherical wave) incidence and farfield (plane wave) incidence are made. [Work supported by NAVSEA Division Newport ILIR Program.]

9:40

3aEA6. Testing facility concepts for the material characterization of porous media consisting of relatively limp foam and stiff fluid. Michael Woodworth and Jeffrey Cipolla (ASI, Weidlinger Assoc., Inc., 1825 K St NW, #350, Washington, DC 20006, michael.woodworth@wai.com)

Fluid filled foams are important components of acoustical systems. Most are made up of a stiff skeleton medium relative to the fluid considered, usually air. Biot's theory of poroelasticity is appropriate for characterizing and modeling these foams. The use of relatively stiff fluid (such as water) and limp foam media pose a greater challenge. Recently modifications to Biot's theory have generated the mechanical relationships required to model these systems. Necessary static material properties for the model can be obtain through *in vacuo* measurement. Frequency dependent properties are more difficult to obtain. Traditional impedance tube methods suffer from fluid structure interaction when the bulk modulus of the fluid media approaches that of the waveguide. The current investigation derives the theory for, and investigates the feasibility of, several rigid impedance tube alternatives for characterizing limp foams in stiff fluid media. Alternatives considered include a sufficiently rigid impedance tube, a pressure relief impedance tube and, the most promising, a piston excited oscillating chamber of small aspect ratio. The chamber concept can recover the descriptive properties of a porous medium described by Biot's theory or by complex-impedance equivalent-fluid models. The advantages of this facility are small facility size, low cost, and small sample size.

10:00–10:20 Break

10:20

3aEA7. Adomian decomposition identifies an approximate analytical solution for a set of coupled strings. David Segala (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, david.segala@navy.mil)

The use of Adomian decomposition method (ADM) has been successfully applied in various applications across the applied mechanics and mathematics community. Originally, Adomian developed this method to derive analytical approximate solutions to nonlinear functional equations. It was shown that the solution to the given nonlinear functional equation can be approximated by an infinite series solution of the linear and nonlinear terms, provided the nonlinear terms are represented by a sum of series of Adomian polynomials. Here, ADM is used to derive an approximate analytical solution to a set of partial differential equations (PDEs) describing the motion of two coupled strings that lie orthogonal to each other. The PDEs are derived using Euler-Lagrange equations of motion. The ends of the strings are pinned and the strings are coupled with a nonlinear spring. A finite element model of the system is developed to provide a comparative baseline. Both the finite element model and analytical solution were driven by an initial displacement condition. The results from both the FEA and analytical solution were compared at six different equally spaced time points over the course of a 1.2 second simulation.

10:40

3aEA8. Comprehensive and practical explorations of nonlinear energy harvesting from stochastic vibrations. Ryan L. Harne and Kon-Well Wang (Mech. Eng., Univ. of Michigan, 2350 Hayward St., 2250 GG Brown Bldg., Ann Arbor, MI 48109-2125, rharne@umich.edu)

Conversion of ambient vibrational energies to electrical power is a recent, popular motivation for research that seeks to realize self-sustaining electronic systems including biomedical implants and remote wireless structural sensors. Many vibration resources are stochastic with spectra concentrated at extremely low frequencies, which is a challenging bandwidth to target in the design of compact, resonant electromechanical harvesters. Exploitation of design-based nonlinearities has uncovered means to reduce and broaden a harvester's frequency range of greatest sensitivity to be more compatible with ambient spectra, thus dramatically improving energy conversion performance. However, studies to date draw differing conclusions regarding the viability of the most promising nonlinear harvesters, namely, those designed around the elastic stability limit, although the investigations present findings having limited verification. To help resolve the outstanding questions about energy harvesting from stochastic vibrations using systems designed near the elastic stability limit, this research integrates rigorous analytical, numerical, and experimental explorations. The harvester architecture considered is a cantilever beam, which is the common focus of contemporary studies, and evaluates critical, practical factors involved for its effective implementation. From the investigations, the most favorable incorporations of nonlinearity are identified and useful design guidelines are proposed.

11:00

3aEA9. Response of infinite length bars and beams with periodically varying area. Andrew J. Hull and Benjamin A. Cray (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil)

This talk develops a solution method for the longitudinal motion of a rod or the flexural motion of a beam of infinite length whose area varies periodically. The conventional rod or beam equation of motion is used with the area and moment of inertia expressed using analytical functions of the longitudinal (horizontal) spatial variable. The displacement field is written as a series expansion using a periodic form for the horizontal wavenumber. The area and moment of inertia expressions are each expanded into a Fourier series. These are inserted into the differential equations of motion and the resulting algebraic equations are orthogonalized to produce a matrix equation whose solution provides the unknown wave propagation coefficients, thus yielding the displacement of the system. An example problem of both a rod and beam are analyzed for three different geometrical shapes. The solutions to both problems are compared to results from finite element analysis for validation. Dispersion curves of the systems are shown graphically. Convergence of the series solutions is illustrated and discussed.

Contributed Papers

11:20

3aEA10. On the exact analytical solutions to equations of nonlinear acoustics. Alexander I. Kozlov (Medical and biological Phys., Vitebsk State Medical Univ., 27, Frunze Ave., Vitebsk 210023, Belarus, albapasserby@yahoo.com)

Some different equations derived as second-order approximations to complete system of equations of nonlinear acoustics of Newtonian media (such as Lighthill-Westerwelt equation, Kuznetsov one, etc.) are usually solved numerically or at least approximately. A general exact analytical method of solution of these problems based on a short chain of changes of variables is presented in the work. It is shown that neither traveling-wave solutions nor classical soliton-like solutions obey these equations. There are three types of possible forms of acoustical pressure depending on parameters of initial equation: so-called continuous shock (or diffusive soliton), a monotonously decaying solution as well as a sectionally continuous periodic one. Obtained results are in good qualitative agreement with previously published numerical calculations of different authors.

11:35

3aEA11. A longitudinal shear wave and transverse compressional wave in solids. ali Zorgani (LabTAU, INSERM, Univ. of Lyon, Bron, France), Stefan Catheline (LabTAU, INSERM, Univ. of Lyon, 151 Cours Albert Thomas, Lyon, France, stefan.catheline@inserm.fr), and Nicolas Benech (Instituto de fisica, Facultad de ciencia, Montevideo, Uruguay)

What general definition can one give to elastic P- and S-wave, especially when they are transversely and longitudinally polarized respectively? This question is the main motivation of the analysis of the Green's function reported in this letter. By separating the Green's function in a divergence free and a rotational free terms, not only a longitudinal S-wave but also a transversal P-wave are described. These waves are shown to be parts of the solution of the wave equation known as coupling terms. Similarly to surface water wave, they are divergence and rotational free. Their special motion is carefully described and illustrated.

Session 3aID

Student Council, Education in Acoustics and Acoustical Oceanography: Graduate Studies in Acoustics (Poster Session)

Zhao Peng, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska-Lincoln, 1110 S. 67th Street, Omaha, NE 68182

Preston S. Wilson, Cochair
Mech. Eng., The University of Texas at Austin, 1 University Station, C2200, Austin, TX 78712

Whitney L. Coyle, Cochair
The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

All posters will be on display from 9:00 a.m. to 11:00 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:00 a.m.

Invited Papers

3aID1. The Graduate Program in Acoustics at The Pennsylvania State University. Victor Sparrow and Daniel A. Russell (Grad. Prog. Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

In 2015, the Graduate Program in Acoustics at Penn State will be celebrating 50 years as the only program in the United States offering the Ph.D. in Acoustics as well as M.S. and M.Eng. degrees in Acoustics. An interdisciplinary program with faculty from a variety of academic disciplines, the Acoustics Program is administratively aligned with the College of Engineering and closely affiliated with the Applied Research Laboratory. The research areas include: ocean acoustics, structural acoustics, signal processing, aeroacoustics, thermoacoustics, architectural acoustics, transducers, computational acoustics, nonlinear acoustics, marine bioacoustics, noise and vibration control, and psychoacoustics. The course offerings include fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound-structure interaction, digital signal processing, experimental techniques, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, ultrasonic NDE, outdoor sound propagation, computational acoustics, flow induced noise, spatial sound and 3D audio, marine bioacoustics, and the acoustics of musical instruments. Penn State Acoustics graduates serve widely throughout military and government labs, academic institutions, consulting firms, and industry. This poster will summarize faculty, research areas, facilities, student demographics, successful graduates, and recent enrollment and employment trends.

3aID2. Graduate studies in acoustics and noise control in the School of Mechanical Engineering at Purdue University. Patricia Davies, J. Stuart Bolton, and Kai Ming Li (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907-2099, daviesp@purdue.edu)

The acoustics community at Purdue University will be described with special emphasis on the graduate program in Mechanical Engineering (ME). Purdue is home to around 30 faculty who study various aspects of acoustics and related disciplines, and so, there are many classes to choose from as graduate students structure their plans of study to complement their research activities and to broaden their understanding of the various aspects of acoustics. In Mechanical Engineering, the primary emphasis is on understanding noise generation, noise propagation, and the impact of noise on people, as well as development of noise control strategies, experimental techniques, and noise and noise impact prediction tools. The ME acoustics research is conducted at the Ray W. Herrick Laboratories, which houses several large acoustics chambers that are designed to facilitate testing of a wide array mechanical systems, reflecting the Laboratories' long history of industry-relevant research. Complementing the acoustics research, Purdue has vibrations, dynamics, and electromechanical systems research programs and is home to a collaborative group of engineering and psychology professors who study human perception and its integration into engineering design. There are also very strong ties between ME acoustics faculty and faculty in Biomedical Engineering and Speech Language and Hearing Sciences.

3aID3. Acoustics program at the University of Rhode Island. Gopu R. Potty, James H. Miller, Brenton Wallin (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Charles E. White (Naval Undersea Warfare Ctr., Newport, RI), and Jennifer Giard (Marine Acoust., Inc., Middletown, RI)

The undergraduate and graduate program in Ocean Engineering at the University of Rhode Island is one of the oldest such programs in the United States. This program offers Bachelors, Masters (thesis and non-thesis options), and Ph.D. degrees in Ocean Engineering. The Ocean Engineering program has a strong acoustic component both at the undergraduate and graduate level. At the graduate level, students can specialize in several areas including geoacoustic inversion, propagation modeling, marine mammal acoustics, ocean acoustic instrumentation, transducers, etc. Current acoustics related research activities of various groups will be presented. Information regarding the requirements of entry into the program will be provided. Many graduates from the program hold faculty positions at a number of universities in the United States and abroad. In addition, graduates from the ocean acoustics program at URI are key staff at many companies and organizations. The opportunities and challenges facing the program will be summarized.

3aID4. Graduate education and research in architectural acoustics at Rensselaer Polytechnic Institute. Ning Xiang, Jonas Braasch, and Todd Brooks (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180, xiangn@rpi.edu)

The rapid pace of change in the fields of architectural-, physical-, and psycho-acoustics has constantly advanced the Graduate Program in Architectural Acoustics from its inception in 1998 with an ambitious mission of educating future experts and leaders in architectural acoustics. Recent years we have reshaped its pedagogy using "STEM" (science, technology, engineering, and mathematics) methods, including intensive, integrative hands-on experimental components that fuse theory and practice in a collaborative environment. Our pedagogy enables graduate students from a broad range of fields to succeed in this rapidly changing field. The graduate program has attracted graduate students from a variety of disciplines including individuals with B.S., B.A., or B.Arch. degrees in Engineering, Physics, Mathematics, Computer Science, Electronic Media, Sound Recording, Music, Architecture, and related fields. RPI's Graduate Program in Architectural Acoustics has since graduated more than 100 graduates with both M.S. and Ph.D. degrees. Along with faculty members they have also actively 0contributed to the program's research in architectural acoustics, psychoaoustics, communication acoustics, signal processing in acoustics as well as our scientific exploration at the intersection of cutting edge research and traditional architecture/music culture. This paper shares the growth and evolution of the graduate program.

3aID5. Graduate training opportunities in the hearing sciences at the University of Louisville. Pavel Zahorik, Jill E. Preminger (Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville School of Medicine, Psychol. and Brain Sci., Life Sci. Bldg. 317, Louisville, KY 40292, pavel.zahorik@louisville.edu), and Christian E. Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

The University of Louisville currently offers two branches of training opportunities for students interested in pursuing graduate training in the hearing sciences: A Ph.D. degree in experimental psychology with concentration in hearing science, and a clinical doctorate in audiology (Au.D.). The Ph.D. degree program offers mentored research training in areas such as psychoacoustics, speech perception, spatial hearing, and multisensory perception, and guarantees students four years of funding (tuition plus stipend). The Au.D. program is a 4-year program designed to provide students with the academic and clinical background necessary to enter audiologic practice. Both programs are affiliated with the Heuser Hearing Institute, which, along with the University of Louisville, provides laboratory facilities and clinical populations for both research and training. An accelerated Au.D./Ph.D. training program that integrates key components of both programs for training of students interested in clinically based research is under development. Additional information is available at http://louisville.edu/medicine/degrees/audiology and http://louisville.edu/psychology/graduate/vision-hearing.

3aID6. Graduate studies in acoustics, Speech and Hearing at the University of South Florida, Department of Communication Sciences and Disorders. Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

This poster will provide an overview of programs and opportunities for students who are interested in learning more about graduate studies in the Department of Communication Sciences and Disorders at the University of South Florida. Ours is a large and active department, offering students the opportunity to pursue either basic or applied research in a variety of areas. Current strengths of the research faculty in the technical areas of Speech Communication and Psychological and Physiological Acoustics include the following: second-language speech perception and production, aging, hearing loss and speech perception, auditory physiology, and voice acoustics and voice quality. Entrance requirements and opportunities for involvement in student research and professional organizations will also be described.

3aID7. Graduate programs in Hearing and Speech Sciences at Vanderbilt University. G. Christopher Stecker and Anna C. Diedesch (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232-8242, g.christopher.stecker@vanderbilt.edu)

The Department of Hearing and Speech Sciences at Vanderbilt University is home to several graduate programs in the areas of Psychological and Physiological Acoustics and Speech Communication. Programs include the Ph.D. in Audiology, Speech-Language Pathology, and Hearing or Speech Science, Doctor of Audiology (Au.D.), and Master's programs in Speech-Language Pathology and Education of the Deaf. The department is closely affiliated with Vanderbilt University's Graduate Program in Neurobiology. Several unique aspects of the research and training environment in the department provide exceptional opportunities for students interested in studying the basic science as well as clinical-translational aspects of auditory function and speech communication in complex environments. These include anechoic and reverberation chambers capable of multichannel presentation, the Dan Maddox Hearing Aid Laboratory, and close connections to active Audiology, Speech-Pathology, Voice, and Otolaryngology clinics. Students interested in the neuroscience of communication utilize laboratories for auditory and multisensory neurophysiology and neuroanatomy, human electrophysiology and neuroimaging housed within the department and at the neighboring Vanderbilt University Institute for Imaging Science. Finally, department faculty and students engage in numerous engineering and industrial collaborations, which benefit from our home within Vanderbilt University and setting in Music City, Nashville. Tennessee.

3aID8. Underwater acoustics graduate study at the Applied Physics Laboratory, University of Washington. Robert I. Odom (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, odom@apl.washington.edu)

With faculty representation in the Departments of Electrical Engineering, and Mechanical Engineering within the College of Engineering, the School of Oceanography, and the Department of Earth and Space Sciences within the College of the Environment, underwater acoustics at APL-UW touches on topics as diverse as long range controlled source acoustics, very low frequency seismics, sediment acoustics, marine mammal vocalizations, and noise generated by industrial activities such as pile driving, among other things. Graduate studies leading to both M.S. and Ph.D. degrees are available. Examples of projects currently being pursued and student opportunities are highlighted in this poster.

3aID9. Graduate acoustics at Brigham Young University. Timothy W. Leishman, Kent L. Gee, Tracianne B. Neilsen, Scott D. Sommerfeldt, Jonathan D. Blotter, and William J. Strong (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

Graduate studies in acoustics at Brigham Young University prepare students for jobs in industry, research, and academia by complementing in-depth coursework with publishable research. In the classroom, a series of five graduate-level core courses provides students with a solid foundation in core acoustics principles and practices. The associated lab work is substantial and provides hands-on experience in diverse areas of acoustics: calibration, directivity, scattering, absorption, Doppler vibrometry, lumped-element mechanical systems, equivalent circuit modeling, arrays, filters, room acoustics measurements, active noise control, and near-field acoustical holography. In addition to coursework, graduate students complete independent research projects with faculty members. Recent thesis and dissertation topics have included active noise control, directivity of acoustic sources, room acoustics, radiation and directivity of musical instruments, energy-based acoustics, aeroacoustics, propagation modeling, nonlinear propagation, and high-amplitude noise analysis. In addition to their individual projects, graduate students often serve as peer mentors to undergraduate students on related projects and often participate in field experiments to gain additional experience. Students are expected to develop their communication skills, present their research at multiple professional meetings, and publish it in peer-reviewed acoustics journals. In the past five years, nearly all graduate students have published at least one refereed paper.

3aID10. Acoustics-related research in the Department of Speech and Hearing Sciences at Indiana University. Tessa Bent, Steven Lulich, Robert Withnell, and William Shofner (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

In the Department of Speech and Hearing Sciences at Indiana University, there are many highly active laboratories that conduct research on a wide range of areas in acoustics. Four of these laboratories are described below. The Biophysics Lab (PI: Robert Withnell) focuses on the mechanics of hearing. Acoustically based signal processing and data acquisition provide experimental data for model-based analysis of peripheral sound processing. The Comparative Perception Lab (PI: William Shofner) focuses on how the physical features of complex sounds are related to their perceptual attributes, particularly pitch and speech. Understanding behavior and perception in animals, particularly in chinchillas, is an essential component of the research. The Speech Production Laboratory (PI: Steven Lulich) conducts research on imaging of the tongue and oral cavity, speech breathing, and acoustic modeling of the whole vocal/respiratory tract. Laboratory equipment includes 3D/4D ultrasound, digitized palate impressions, whole-body and inductive plethysmography, electroglottography, oral and nasal pressure and flow recordings, and accelerometers. The Speech Perception Lab (PI: Tessa Bent) focuses on the perceptual consequences of phonetic variability in speech, particularly foreign-accented speech. The main topics under investigation are perceptual adaptation, individual differences in word recognition, and developmental speech perception.

3aID11. Biomedical research at the image-guided ultrasound therapeutics laboratories. Christy K. Holland (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3935, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu), T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Kevin J. Haworth, Kenneth B. Bader, Himanshu Shekhar, and Kirthi Radhakrishnan (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

The Image-guided Ultrasound Therapeutic Laboratories (IgUTL) are located at the University of Cincinnati in the Heart, Lung, and Vascular Institute, a key component of efforts to align the UC College of Medicine and UC Health research, education, and clinical programs. These extramurally funded laboratories, directed by Prof. Christy K. Holland, are comprised of graduate and undergraduate students, postdoctoral fellows, principal investigators, and physician-scientists with backgrounds in physics and biomedical engineering, and clinical and scientific collaborators in fields including cardiology, neurosurgery, neurology, and emergency medicine. Prof. Holland's research focuses on biomedical ultrasound including sonothrombolysis, ultrasound-mediated drug and bioactive gas delivery, development of echogenic liposomes, early detection of cardiovascular diseases, and ultrasound-image guided tissue ablation. The Biomedical Ultrasonics and Cavitation Laboratory within IgUTL, directed by Prof. Kevin J. Haworth, employs ultrasound-triggered phase-shift emulsions (UPEs) for image-guided treatment of cardiovascular disease, especially thrombotic disease. Imaging algorithms incorporate both passive and active cavitation detection. The Biomedical Acoustics Laboratory within IgUTL, directed by Prof. T. Douglas Mast, employs ultrasound for monitoring thermal therapy, ablation of cancer and vascular targets, transdermal drug delivery, and noninvasive measurement of tissue deformation.

3aID12. Graduate acoustics education in the Cockrell School of Engineering at The University of Texas at Austin. Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Neal A. Hall (Elec. and Comp. Eng. Dept., The Univ. of Texas at Austin, Austin, TX), Mark F. Hamilton (Mech. Eng. Dept., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712), Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept., The Univ. of Texas at Austin, Austin, TX, pswilson@mail.utexas.edu)

While graduate study in acoustics takes place in several colleges and schools at The University of Texas at Austin (UT Austin), including Communication, Fine Arts, Geosciences, and Natural Sciences, this poster focuses on the acoustics program in Engineering. The core of this program resides in the Departments of Mechanical Engineering (ME) and Electrical and Computer Engineering (ECE). Acoustics faculty in each department supervise graduate students in both departments. One undergraduate and seven graduate acoustics courses are cross-listed in ME and ECE. Instructors for these courses include staff at Applied Research Laboratories at UT Austin, where many of the graduate students have research assistantships. The undergraduate course, taught every fall, begins with basic physical acoustics and proceeds to draw examples from different areas of engineering acoustics. Three of the graduate courses are taught every year: a two-course sequence on physical acoustics, and a transducers course. The remaining four graduate acoustics courses, taught in alternate years, are on nonlinear acoustics, underwater acoustics, ultrasonics, and architectural acoustics. An acoustics seminar is held most Fridays during the long semesters, averaging over ten per semester since 1984. The ME and ECE departments both offer Ph.D. qualifying exams in acoustics.

3aID13. Graduate studies in Ocean Acoustics in the Massachusetts Institute of Technology and Woods Hole Oceanographic Institution Joint Program. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

An overview of graduate studies in Ocean Acoustics within the framework of the Massachusetts Institute of Technology (MIT) and Woods Hole Oceanographic Institution (WHOI) Joint Program is presented, including a brief history of the program, facilities, details of the courses offered, alumni placing, funding opportunities, and current program status, faculty members and research. Emphasis is given to the key role of the joint strengths provided by MIT and WHOI, the strong sea-going history of the program, and the potential for highly interdisciplinary research.

3aID14. Graduate studies in acoustics at the University of Notre Dame. Christopher Jasinski and Thomas C. Corke (Aerosp. and Mech. Eng., Univ. of Notre Dame, 54162 Ironwood Rd., South Bend, IN 46635, chrismjasinski@gmail.com)

The University of Notre Dame department of Aerospace and Mechanical Engineering is conducting cutting edge research in aeroacoustics, structural vibration, and wind turbine noise. Expanding facilities are housed at two buildings of the Hessert Laboratory for Aerospace Engineering and include two 25 kW wind turbines, a Mach 0.6 wind tunnel, and an anechoic wind tunnel. Several faculty members conduct research related to acoustics and multiple graduate level courses are offered in general acoustics and aeroacoustics. This poster presentation will give an overview of the current research activities, laboratory facilities, and graduate students and faculty involved at Notre Dame's Hessert Laboratory for Aerospace Engineering.

3aID15. Graduate study in Architectural Acoustics within the Durham School at the University of Nebraska—Lincoln. Lily M. Wang, Matthew G. Blevins, Zhao Peng, Hyun Hong, and Joonhee Lee (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 South 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

Persons interested in pursuing graduate study in architectural acoustics are encouraged to consider joining the Architectural Engineering Program within the Durham School of Architectural Engineering and Construction at the University of Nebraska—Lincoln (UNL). Among the 21 ABET-accredited Architectural Engineering (AE) programs across the United States, the Durham School's program is one of the few that offers graduate engineering degree programs (MAE, MS, and PhD) and one of only two that offers an area of concentration in architectural acoustics. Acoustics students in the Durham School benefit both from the multidisciplinary environment in an AE program and from our particularly strong ties to the building industry, since three of the largest architectural engineering companies in the United States are headquartered in Omaha, Nebraska. Descriptions will be given on the graduate-level acoustics courses, newly renovated acoustic lab facilities, the research interests and achievements of our acoustics faculty and students, and where our graduates are to date. Our group is also active in extracurricular activities, particularly through the University of Nebraska Acoustical Society of America Student Chapter. More information on the "Nebraska Acoustics Group" at the Durham School may be found online at http://nebraskaacousticsgroup.org/.

3aID16. Pursuing the M.Eng. in acoustics through distance education from Penn State. Daniel A. Russell and Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, drussell@engr.psu.edu)

Since 1987, the Graduate Program in Acoustics at Penn State has been providing remote access to graduate level education leading to the M.Eng. degree in Acoustics. Course lecture content is currently broadcast as a live-stream via Adobe Connect to distance students scattered throughout North America and around the world, while archived recordings allow distance students to access lecture material at their convenience. Distance Education students earn the M.Eng. in Acoustics degree by completing 30 credits of coursework (six required core courses and four electives) and writing a capstone paper. Courses offered for distance education students include: fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, aerodynamic noise, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, flow induced noise, spatial sound

and 3D audio, marine bioacoustics, and acoustics of musical instruments. This poster will summarize the distance education experience leading to the M.Eng. degree in Acoustics from Penn State showcasing student demographics, capstone paper topics, enrollment statistics and trends, and the success of our graduates.

3aID17. Graduate studies in acoustics at Northwestern University. Ann Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL IL, abradlow@northwestern.edu)

Northwestern University has a vibrant and highly interdisciplinary community of acousticians. Of the 13 ASA technical areas, three have strong representation at Northwestern: Speech Communication, Psychological and Physiological Acoustics, and Musical Acoustics. Sound-related work is conducted across a wide range of departments including Linguistics (in the Weinberg College of Arts and Sciences), Communication Sciences & Disorders, and Radio/Television/Film (both in the School of Communication), Electrical Engineering & Computer Science (in the McCormick School of Engineering), Music Theory & Cognition (in the Bienen School of Music), and Otolaryngology (in the Feinberg School of Medicine). In addition, The Knowles Hearing Center involves researchers and labs across the university dedicated to the prevention, diagnosis and treatment of hearing disorders. Specific acoustics research topics across the university range from speech perception and production across the lifespan and across languages, dialect and socio-indexical properties of speech, sound design, machine perception of music and audio, musical communication, the impact of long-term musical experience on auditory encoding and representation, auditory perceptual learning, and the cellular, molecular, and genetic bases of hearing function. We invite you to visit our poster to learn more about the "sonic boom" at Northwestern University!

WEDNESDAY MORNING, 29 OCTOBER 2014 SANTA FE, 9:00 A.M. TO 11:45 A.M.

Session 3aMU

Musical Acoustics: Topics in Musical Acoustics

Jack Dostal, Chair
Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, NC 27109

Contributed Papers

9:00

3aMU1. Study of free reed attack transients using high speed video. Spencer Henessee (Phys., Coe College, GMU #447, 1220 First Ave. NE, Cedar Rapids, IA 52402, sahenessee@coe.edu), Daniel M. Wolff (Univ. of North Carolina at Greensboro, Greensboro, NC), and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

Earlier methods of studying the motion of free reeds have been augmented with the use of high-speed video, resulting in a more detailed picture of reed oscillation, especially the initial transients. Displacement waveforms of selected points on the reed tongue image can be obtained using appropriate tracking software. The waveforms can be analyzed for the presence of higher modes of vibration and other features of interest in reed oscillation, and they can be used in conjunction with displacement or velocity waveforms obtained by other means, along with finite element simulations, to obtain detailed information about reed oscillation. The high speed video data has a number of advantages. It can provide a two-dimensional image of the motion of any point tracked on the reed tongue, and the freedom to change the points selected for tracking provides flexibility in data acquisition. In addition, the high speed camera is capable of simultaneous triggering of other motion sensors as well as oscilloscopes and spectrum analyzers. Some examples of the use of high speed video are presented and some difficulties in the use of this technique are discussed. [Work partially supported by US National Science Foundation REU Grant PHY-1004860.]

9:15

3aMU2. Detailed analysis of free reed initial transients. Daniel M. Wolff (Univ. of North Carolina at Greensboro, 211 McIver St. Apt. D, Greensboro, NC 27403, dmwolff@uncg.edu), Spencer Henessee, and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

The motion of the reed tongue in early stages of the attack transient has been studied in some detail for reeds from a reed organ. Oscillation waveforms were obtained using a laser vibrometer system, variable impedance transducer proximity sensors, and high speed video with tracking software. Typically, the motion of the reed tongue begins with an initial displacement of the equilibrium position, often accompanied by a few cycles of irregular oscillation. This is followed by a short transitional period in which the amplitude of oscillation gradually increases and the frequency stabilizes at the steady state oscillation frequency. In the next stage, the amplitude of oscillation continues to increase to the steady state value. The spectra derived from the waveforms in each stage have been analyzed, showing that the second transverse mode and the first torsional mode are both observed in the transient, with the amplitude of the torsional mode apparently especially significant in the earlier stages of oscillation. Comparison of reed tongues of different design have been made to explore the role of the torsional mode the initial excitation. Finite element simulations have been used to aid in the verification and interpretation of some of the results. [Work supported by US National Science Foundation REU Grant PHY-1004860.]

9:30

3aMU3. Comparison of traditional and matched grips: Rhythmic sequences played in jazz drumming. E. K. Ellington Scott (Oberlin College, OCMR2639, Oberlin College, Oberlin, OH 44074, escott@oberlin.edu) and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

Traditional and matched grips have been compared using a series of measurements involving rhythmic sequences played by experienced jazz drummers using each of the two grips. Rhythmic sequences played on the snare drum were analyzed using high speed video as well as other measurement techniques including laser vibrometry and spectral analysis of the sound waveforms. The high speed video images, used with tracking software, allow observation of several aspects of stick-drum head interaction. These include two-dimensional trajectories of the drum stick tip, a detailed picture of the stick-drum head interaction, and velocities of both the stick and the drum head during the contact phase of the stroke. Differences between the two grips in timing during the rhythmic sequences were investigated, and differences in sound spectrum were also analyzed. Some factors that may be player dependent have been explored, such as the effect of tightness of the grip, but an effort has been made to concentrate on factors that are independent of the player. [Work supported by US National Science Foundation REU Grant PHY-1004860.]

9:45

3aMU4. A harmonic analysis of oboe reeds. Julia Gjebic, Karen Gipson (Phys., Grand Valley State Univ., 10255 42nd Ave., Apt. 3212, Allendale, MI 49401, gjebicj@mail.gvsu.edu), and Marlen Vavrikova (Music and Dance, Grand Valley State Univ., Allendale, MI)

Because oboists make their own reeds to satisfy personal and physiological preferences, no two reed-makers construct their reeds in the same manner, just as no two oboe players have the same sound. The basic structure of an oboe reed consists of two curved blades of the grass Arundo donax bound to a conical metal tube (a staple) such that the edges of the blades meet and vibrate against one another when stimulated by a change in the surrounding pressure. While this basic structure is constant across reed-makers, the physical measurements of the various portions of the reed (tip, spine, and heart) resulting from the final stage of reed-making (scraping) can vary significantly between individual oboists. In this study, we investigated how the physical structure of individual reeds relates to the acoustic spectrum. We performed statistical analyses to discern which areas of the finished reed influence the harmonic series most strongly. This information is of great interest to oboists as it allows them quantitative insight into how their individual scrape affects their overall tone quality and timbre.

10:00

3aMU5. Modeling and numerical simulation of a harpsichord. Rossitza Piperkova, Sebastian Reiter, Martin Rupp, and Gabriel Wittum (Goethe Ctr. for Sci. Computing, Goethe Univ. Frankfurt, Kettenhofweg 139, Frankfurt am Main 60325, Germany, Wittum@gcsc.uni-frankfurt.de)

This research studies what influences various properties of a soundboard may have upon the acoustic feedback to gain a better understanding about the relevance of different properties regarding the sound characteristics. It may also help to improve the quality of simulations. We did a modal analysis of a real soundboard of a harpsichord using a Laser-Doppler-Vibrometer and also simulated several models of the very same soundboard in three space dimensions using the simulation software UG4. The used models of the sound board differed from each other by changing or skipping several properties and components. Then, we compared the simulated vibration patterns with the patterns measured on the real sound board to gain a better understanding about their influences on the vibrations. In particular, we used models with and without soundboard bars and bridge, but also were using different thicknesses for the soundboard itself.

10:15–10:30 Break

10:30

3aMU6. Temporal analysis, manipulation, and resynthesis of musical vibrato. Mingfeng Zhang, Gang Ren, Mark Bocko (Dept. Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, mzhang43@hse.rochester.edu), and James Beauchamp (Dept. Elec. and Comput. Eng., Univ. of Illinois at Urbana–Champaign, Urbana, IL)

Vibrato is an important music performance technique for both voice and various music instruments. In this paper, a signal processing framework for vibrato analysis, manipulation and resynthesis is presented. In the analysis part, music vibrato is treated as a generalized descriptor of music timbre and the signal magnitude and instantaneous frequency is implemented as temporal features. Specifically, the magnitude track shows the dynamic variations of audio loudness, and the frequency track shows the frequency deviations varying with time. In the manipulation part, several manipulation methods for the magnitude track and the frequency track is implemented. The tracking results are manipulated in both the time- and the frequency-domain. These manipulation methods are implemented as an interactive process to allow musicians to manually adjust the processing parameters. In the resynthesis part, the simulated vibrato audio is created using sinusoidal resynthesis process. The resynthesis part serves three purpose: to imitate human music performance, to migrate sonic features across music performances, and to serve as creative audio design tools, e.g., to create non-existing vibrato characteristics. The source audio from human music performance and the resynthesize audio is compared using subjective listening tests to validate our proposed framework.

10:45

3aMU7. Shaping musical vibratos using multi-modal pedagogical interactions. Mingfeng Zhang, Fangyu Ke (Dept. Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, mzhang43@hse.rochester.edu), James Beauchamp (Dept. Elec. and Comput. Eng., Univ. of Illinois at Urbana–Champaign, Urbana, IL), and Mark Bocko (Dept. Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

The music vibrato is termed a “pulsation in pitch, intensity, and timbre” because of its effectiveness in artistic rendering. However, this sonic trick is largely still a challenge in music pedagogy across music conservatories. In classroom practice, music teachers use demonstration, body gestures, and metaphors to convey their artistic intentions and the modern computer tools are seldom employed. In our proposed framework, we use musical vibrato visualization and sonification tools as a multi-modal computer interface for pedagogical purpose. Specifically, we compare master performance audio with student performance audio using signal analysis tools. Then, we obtain various similarity measures based on these signal analysis results. Based on these similarity measures we implement multi-modal interactions for music students to shape their music learning process. The visualization interface is based on audio features including dynamics, pitch and timbre. The sonifications interface is based on recorded audio and synthesized audio. To enhance the music relevance of our proposed framework, both visualization and sonification tools are targeted to serve a musical communicating to convey musical concepts in an intuitive manner. The proposed framework is evaluation using subjective ratings from music students and objective assessment of measurable training goals.

11:00

3aMU8. Absolute memory for popular songs is predicted by auditory working memory ability. Stephen C. Van Hedger, Shannon L. Heald, Rachelle Koch, Howard C. Nusbaum (Psych., The Univ. of Chicago, 5848 S. University Ave., Beecher 406, Chicago, IL 60637, stephen.c.hedger@gmail.com),

While most individuals do not possess absolute pitch (AP)—the ability to name an isolated musical note in absence of a reference note—they do show some limited memory for absolute pitch of melodies. For example, most individuals are able to recognize when a well-known song has been subtly pitch shifted. Presumably, individuals are able to select the correct absolute pitch at above-chance levels because well-known songs are frequently heard at a consistent pitch. In the current studies, we ask whether individual differences in absolute pitch judgments for people without AP can be explained by general differences in auditory working memory.

Working memory capacity has been shown to predict the perceptual fidelity of long-term category representations in vision; thus, it is possible that auditory working memory capacity explains individual differences in recognizing the tuning of familiar songs. We found that participants were reliably above chance in classifying popular songs as belonging to the correct or incorrect key. Moreover, individual differences in this recognition performance were predicted by auditory working memory capacity, even after controlling for overall music experience and stimulus familiarity. Implications for the interaction between working memory and AP are discussed.

11:15

3aMU9. Constructing alto saxophone multiphonic space. Keith A. Moore (Music, Columbia Univ., 805 W Church St., Savoy, Illinois 10033, kam101@columbia.edu)

Multiphonics are sonorities with two or more independent tones arising from instruments, or portions of instruments, associated with the production of single pitches. Since the 1960s multiphonics have been probed in two ways. Acousticians have explored the role of nonlinearity in multiphonic sound production (Benade 1976; Backus 1978; Keefe & Laden 1991) and musicians have created instrumental catalogs of multiphonic sounds (Bartolozzi 1967; Rehfeldt 1977; Kientzy 1982; Levine 2002). These lines of inquiry have at times been combined (Veale & Mankopf 1994). However, a meta-level analysis has not yet emerged from this work that answers basic questions such as how many kinds of multiphonics are found on one particular instrument and which physical conditions underlie such variety. The present paper suggests a database driven approach to the problem, producing a "quantitative resonant frequency curve" that shows every audible appearance of each frequency in a large—if not permutationally exhaustive—set of alto saxophone multiphonics. Compelling data emerges, including sonority prototypes, prototype transposition levels, and register specific distortions. Notably, true difference tones—audible difference tones unsustainable apart from a sounding multiphonic—are found to be register specific, not sonority specific; suggesting that physical locations (rather than harmonic contexts) underpin these sounds.

11:30

3aMU10. Linear-response reflection coefficient of the recorder air-jet amplifier. John C. Price (Phys., Univ. of Colorado, 390 UCB, Boulder, CO 80309, john.price@colorado.edu), William Johnston (Phys., Colorado State Univ., Fort Collins, CO), and Daniel McKinnon (Chemical Eng., Univ. of Colorado, Boulder, CO)

Steady-state oscillations in a duct flute, such as the recorder, are controlled by (1) closing tone holes and (2) adjusting the blowing pressure or air-jet velocity. The acoustic amplitude in steady-state cannot be controlled independent of the jet velocity, because it is determined by the gain saturation properties of the air-jet amplifier. Consequently, the linear-response gain of the air-jet amplifier has only very rarely been studied [Thwaites and Fletcher, J. Acoust. Soc. Am. **74**, 400–408 (1983)]. Efforts have focused instead on the more complex gain-saturated behavior, which is controlled by vortex shedding at the labium. We replace the body of a Yamaha YRT-304B tenor recorder with a multi-microphone reflectometer and measure the complex reflection coefficient of the head at small acoustic amplitudes as a function of air-jet velocity and acoustic frequency. We find that the gain (reflection coefficient magnitude) has a maximum value of 2.5 at a Strouhal number of $\approx$0.3 (jet transit time divided by acoustic period), independent of jet velocity. Surprisingly, the frequency where the gain peaks for a given blowing pressure is not close to the in-tune pitch of a note that is played at the same blowing pressure.

WEDNESDAY MORNING, 29 OCTOBER 2014 MARRIOTT 3/4, 8:45 A.M. TO 12:00 NOON

Session 3aNS

Noise and ASA Committee on Standards: Wind Turbine Noise

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118

Robert D. Hellweg, Cochair
Hellweg Acoustics, 13 Pine Tree Rd., Wellesley, MA 02482

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Drive, Champaign, IL 61821

Kenneth Kaliski, Cochair
RSG Inc., 55 Railroad Row, White River Junction, VT 05001

Invited Papers

8:45

3aNS1. Massachusetts Wind Turbine Acoustics Research Project—Goals and preliminary results. Kenneth Kaliski, David Lozupone (RSG Inc., 55 RailRd. Row, White River Junction, VT 05001, ken.kaliski@rsginc.com), Peter McPhee (Massachusetts Clean Energy Ctr., Boston, MA), Robert O'Neal (Epsilon Assoc., Maynard, MA), John Zimmerman (Northeast Wind, Waterbury, VT), Kieth Wilson (Keith Wilson, Hanover, NH), and Carol Rowan-West (Massachusetts Dept. of Environ. Protection, Boston, MA)

The Commonwealth of Massachusetts (USA) has 43 operating wind turbine projects of 100 kW or more. At several of these projects, noise complaints have been made to state authorities. The Massachusetts Clean Energy Center, which provides funding for early stage analysis and development of wind power projects, and the Massachusetts Department of Environmental Protection, which regulates noise, launched the project to increase understanding of (1) wind turbine acoustic impacts, taking into account variables such as wind turbine size, technology, wind speed, topography and distance, and (2) the generation, propagation, and measurement of sound around wind turbine projects, to inform policy-makers on how pre- and post-construction wind turbine noise studies should be conducted. This study involved the collection of detailed sound and meteorological data at five locations. The resulting database and interim reports contain information on infrasound and audible frequencies, including amplitude modulation, tonality, and level. Analyses will include how the effects of wind shear and other variables may affect these parameters. Preliminary findings reflect the effects of meteorological conditions on wind turbine sound generation and propagation.

9:05

3aNS2. Wind turbine annoyance—A clue from acoustic room modes. William K. Palmer (TRI-LEA-EM, 76 SideRd. 33-34 Saugeen, RR 5, Paisley, ON N0G2N0, Canada, trileaem@bmts.com)

When one admits that they do not know all the answers and sets out to listen to the stories of people annoyed by wind turbines, the clues can seem confusing. Why would some people report that they could get a better night's sleep in an outdoor tent, rather than their bedroom? Others reported that they could sleep better in the basement recreation room of their home, than in bedrooms. That made little sense either. A third mysterious clue came from acoustic measurements at homes nearby wind turbines. Analysis of the sound signature revealed low frequency spikes, but at amplitudes well below those expected to cause annoyance. The clues merged while studying the acoustic room modes in a home, to reveal a remarkable hypothesis as to the cause of annoyance from wind turbines. In rooms where annoyance was felt, the frequencies flagged by room mode calculations and the low frequency spikes observed from wind turbine measurements coincided. This paper will discuss the research and the results, which revealed a finding that provides a clue to the annoyance, and potentially even a manner of providing limited relief.

9:25

3aNS3. A perspective on wind farm complaints and the Acoustical Society of America's public policy. Paul D. Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com) and George Hessler (Hessler Assoc., Haymarket, VA)

Worldwide, hundreds of wind farms have been built and commissioned. A sizeable fraction of these have had some complaints about wind farm noise, perhaps 10 to 50%. A smaller percentage of wind farms have engendered more widespread complaints and claims of adverse health effects, perhaps 1 to 10%. And in the limit (0 to 1%), there have been very widespread, vociferous complaints and in some cases people have abandoned their houses. Some advocates for potentially affected communities have opined that many will be made ill, while living miles from the nearest turbine, and some, who are wind power advocates, have opined that there is no possibility anyone can possibly be made ill from wind turbine acoustic emissions. In an attempt to ameliorate this frequently polarized situation, the ASA has established a public policy statement that calls for the development of a balanced research agenda to establish facts, where "balanced" means the research should resolve issues for all parties with a material interest, and all parties should have a seat at the table where the research plans are developed. This paper presents some thoughts and suggestions as to how this ASA public policy statement can be nurtured and brought to fruition.

9:45

3aNS4. Balancing the research approach on wind turbine effects through improving psychological factors that affect community response. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

There is a substantial need to find a balanced approach to deal with people's concern about wind turbine effects. Indeed, the psychological factors that affect community response will be an important facet in this complete agenda development. Many of these relevant issues are related to the soundscape concept which was adopted as an approach to provide a more holistic evaluation of "noise" and its effects on the quality of life. Moreover, the soundscape technique uses a variety of investigation techniques, taxonomy and measurement methods. This is a necessary protocol to approach a subject or phenomenon, to improve the validity of the research or design outcome and to reduce the uncertainty of relying only on one approach. This presentation will use recent data improving the understanding about the role of psychoacoustic parameters going beyond equivalent continuous sound level in wind turbine affects in order to discuss relevant psychological factors based on soundscape techniques.

10:05–10:25 Break

Contributed Papers

10:25

3aNS5. Measurement and synthesis of wind turbine infrasound. Bruce E. Walker (Channel Islands Acoust., 676 W Highland Dr., Camarillo, CA 93010, noiseybw@aol.com) and Joseph W. Celano (Newson-Brown Acoust. LLC, Santa Monica, CA)

As part of an ongoing investigation into the putative subjective effects of sub-20 Hz acoustical emissions from large industrial wind turbines, measurement techniques for faithful capture of emissions waveforms have been developed and reported. To evaluate perception thresholds, Fourier synthesis and high fidelity low-frequency playback equipment has been used to duplicate in a residential-like listening environment the amplitudes and wave slopes of the actual emissions, with pulsation rate in the range of 0.5–1.0 per second. Further, the amplitudes and slopes of the synthesized waves can be parametrically varied and the harmonic phases "scrambled" to assess the relative effects on auditory and other subjective responses. Measurement and synthesis system details and initial subjective response results will be shown.

10:40

3aNS6. Propagation of wind turbine noise through the turbulent atmosphere. Yuan Peng, Nina Zhou, Jun Chen, and Kai Ming Li (Mech. Eng., Purdue Univ., 177 South Russel St., West Lafayette, IN 47907-2099, peng45@purdue.edu)

It is well known that turbulence can cause fluctuations in the resulting sound fields. In the issue of wind turbine noise, such effect is non-negligible since either the inflow turbulence from nearby turbine wakes or the atmospheric turbulence generated by rotating turbine blades can increase the sound output of individual turbines. In this study, a combined approach of the Finite Element Method (FEM) and Parabolic Equation (PE) method is employed to predict the sound levels from a wind turbine. In the prediction procedure, the near field acoustic data is obtained by means of a computational fluid dynamic program which serves as a good starting field of sound propagation. It is then possible to advance wind turbine noise in range by using the FEM/PE marching algorithm. By incorporating the simulated turbulence profiles near wind turbine, more accurate predictions of sound field in realistic atmospheric conditions are obtained.

10:55–12:00 Panel Discussion

WEDNESDAY MORNING, 29 OCTOBER 2014 INDIANA C/D, 8:20 A.M. TO 11:30 A.M.

Session 3aPA

Physical Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Noise: Acoustics of Pile Driving: Models, Measurements, and Mitigation

Kevin M. Lee, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Mark S. Wochner, Cochair
AdBmTechnologies, 1605 McKinley Ave., Austin, TX 78702

Invited Papers

8:20

3aPA1. Understanding effects of man-made sound on fishes and turtles: Gaps and guidelines. Arthur N. Popper (Biology, Univ. of Maryland, Biology/Psych. Bldg., College Park, MD 20742, apopper@umd.edu) and Anthony D. Hawkins (Loughine Ltd, Aberdeen, United Kingdom)

Mitigating measures may be needed to protect animals and humans that are exposed to sound from man-made sources. In this context, the levels of man-made sound that will disrupt behavior or physically harm the receiver should drive the degree of mitigation that is needed. If a particular sound does not affect an animal adversely, then there is no need for mitigation! The problem then is to know the sound levels that can affect the receiving animal. For most marine animals, there are relatively few data to develop guidelines that can help formulate the levels at which mitigation is needed. In this talk, we will review recent guidelines for fishes and turtles. Since so much remains to be determined in order to make guidelines more useful, it is important that priorities be set for future research. The most critical data, with broadest implications for marine life, should be obtained first. This paper will also consider the most critical gaps and present recommendations for future research.

8:40

3aPA2. The relationship between underwater sounds generated by pile driving and fish physiological responses. Michele B. Halvorsen (CSA Ocean Sci. Inc., 8502 SW Kanner Hwy, Stuart, FL 334997, mhalvorsen@conshelf.com)

Assessment of fish physiology after exposure to impulsive sound has been limited by quantifying physiological injuries, which range from mortal to recoverable. A complex panel of injuries was reduced to a single metric by a model called the Fish Index of Trauma. Over several years, six species of fishes from different morphological groupings, (e.g., physoclistous, physostomous, and lack of a swim bladder) were studied. The onset of physiological tissue effect was determined across a range of cumulative sound exposure levels with varying number of pile strikes. Follow up studies included investigation of healing from incurred injuries. The level of injury that animals expressed was influenced by their morphological grouping. Finally, investigation of the inner ear sensory hair cells showed that damage occurred at higher sound exposure levels than when the onset of tissue injury would occur.

9:00

3aPA3. A model to predict tissue damage in fishes from vibratory and impact pile driving. Mardi C. Hastings (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Predicting effects of underwater pile driving on marine life requires coupling of pile source models with biological receiver models. Fishes in particular are very vulnerable to tissue damage and hearing loss from pile driving activities, especially since they are often restricted to specific habitat sites and migratory routes. Cumulative sound exposure level is the metric used by government agencies for sound exposure criteria to protect marine animals. In recent laboratory studies, physical injury and hearing loss in fish from simulated impact pile driving signals have even been correlated with this metric. Mechanisms for injury and hearing loss in fishes, however, depend on relative acoustic particle motion within the body of the animal, which can be disproportionately large in the vicinity of a pile. Modeling results will be presented showing correlation of auditory tissue damage in three species of fish with relative particle motion that can be generated 10–20 m from driving a 24-in diameter steel pile with an impact hammer. Comparative results with vibratory piling based on measured waveforms indicate that particle motion mechanisms may provide an explanation why the very large cumulative sound exposure levels associated with vibratory pile driving do not produce tissue damage.

9:20

3aPA4. Pile driving pressure and particle velocity at the seabed: Quantifying effects on crustaceans and groundfish. James H. Miller, Gopu R. Potty, and Hui-Kwan Kim (Ocean Eng., Univ. of Rhode Island, URI Bay Campus, 215 South Ferry Rd., Narragansett, RI 02882, miller@egr.uri.edu)

In the United States, offshore wind farms are being planned and construction could begin in the near future along the East Coast of the US. Some of the sites being considered are known to be habitat for crustaceans such as the American lobster, *Homarus americanus,* which has a range from New Jersey to Labrador along the coast of North America. Groundfish such as summer flounder, *Paralichthys dentatus,* and winter flounder, *Pseudopleuronectes americanus,* also are common along the East Coast of the US. Besides sharing the seafloor in locations where wind farms are planned, all three of these species are valuable commercially. We model the effects on crustaceans, groundfish, and other animals near the seafloor due to pile driving. Three different waves are investigated including the compressional wave, shear wave and interface wave. A Finite Element (FE) technique is employed in and around the pile while a Parabolic Equation (PE) code is used to predict propagation at long ranges from the pile. Pressure, particle displacement, and particle velocity are presented as a function of range at the seafloor for a shallow water environment near Rhode Island. We will discuss the potential effects on animals near the seafloor.

9:40

3aPA5. Finite difference computational modeling of marine impact pile driving. Alexander O. MacGillivray (JASCO Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, alex@jasco.com)

Computational models based on the finite difference (FD) method can be successfully used to predict underwater pressure waves generated by marine impact pile driving. FD-based models typically discretize the equations of motion for a cylindrical shell to model the vibrations of a submerged pile in the time-domain. However, because the dynamics of a driven pile are complex, realistic models must also incorporate physics of the driving hammer and surrounding acousto-elastic media into the FD formulation. This paper discusses several of the different physical phenomena involved, and shows some approaches to simulating them using the FD method. Topics include dynamics of the hammer and its coupling to the pile head, transmission of axial pile vibrations into the soil, energy dissipation at the pile wall due to friction, acousto-elastic coupling to the surrounding media, and near-field versus far-field propagation modeling. Furthermore, this paper considers the physical parameters required for predictive modeling of pile driving noise in conjunction with some practical considerations about how to determine these parameters for real-world scenarios.

10:00–10:20 Break

10:20

3aPA6. On the challenges of validating a profound pile driving noise model. Marcel Ruhnau, Tristan Lippert, Kristof Heitmann, Stephan Lippert, and Otto von Estorff (Inst. of Modelling and Computation, Hamburg Univ. of Technol., Denickestraße 17, Hamburg, Hamburg 21073, Germany, mub@tuhh.de)

When predicting underwater sound levels for offshore pile driving by using numerical simulation models, appropriate model validation becomes of major importance. In fact, different parallel transmission paths for sound emission into the water column, i.e., pile-to-water, pile-to-soil, and soil-to-water, make a validation at each of the involved interfaces necessary. As the offshore environment comes with difficult and often unpredictable conditions, measurement campaigns are very time consuming and cost intensive. Model developers have to keep in mind that even thorough planning cannot overcome practical restrictions as well as technical limits and thus require for a reasonable model balancing. The current work presents the validation approach chosen for a comprehensive pile driving noise model—consisting of a near field finite element model as well as a far field propagation model—that is used for the prediction of noise levels at offshore wind farms.

10:40

3aPA7. Underwater noise and transmission loss from vibratory pile driving. Peter H. Dahl and Dara M. Farrell (Appl. Phys. Lab. and Mech. Eng. Dept., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu)

High levels of underwater sound can be produced in vibratory pile driving that can carry regulatory implications. In this presentation, observations of underwater noise from vibratory pile driving made with a vertical line array placed at range 17 m from the source (water depth 7.5 m) are discussed, along with simultaneous measurements made at ranges of order 100 m. It is shown that the dominant spectral

features are related to the frequency of the vibratory pile driving hammer (typically 15–35 Hz), producing spectral lines at intervals of this frequency. Homomorphic analysis removes these lines to reveal the underlying variance spectrum. The mean square pressure versus depth is subsequently studied in octave bands in view of the aforementioned spectral line property, with depth variation well modeled by an incoherent sum of sources distributed over the water column. Adiabatic mode theory is used to model the range dependent local bathymetry, including the effect of elastic seabed, and comparisons are made with simultaneous measurements of the mean-square acoustic pressure at ranges 200 and 400 m. This approach makes clear headway into the problem of predicting transmission loss versus range for this method of pile driving.

Contributed Papers

11:00

3aPA8. Using arrays of air-filled resonators to attenuate low frequency underwater sound. Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Preston S. Wilson (Mech. Eng. Dept. 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 underwater air-filled resonators that could potentially be used in an underwater noise abatement system. The resonators are similar to Helmholtz resonators without a neck, consisting of underwater inverted air-filled cavities with combinations of rigid and elastic wall members, and they are intended to be fastened to a framework to form a stationary array surrounding a noise source, such as a marine pile driving operation, a natural resource production platform, or an air gun array, or to protect a receiving area from outside noise. Previous work has demonstrated the potential of surrounding low frequency sound sources with arrays of large stationary encapsulated bubbles that can be designed to attenuate sound levels over any desired frequency band and with levels of reduction up to 50 dB [Lee and Wilson, Proceedings of Meeting on Acoustics **19**, 075048 (2013)]. Open water measurements of underwater sound attenuation using resonators were obtained during a set of lake experiments, where a low-frequency electromechanical sound source was surrounded by different arrays of resonators. The results indicate that air-filled resonators are a potential alternative to using encapsulated bubbles for low frequency underwater noise mitigation. [Work supported by AdBm Technologies.]

11:15

3aPA9. Axial impact driven buckling dynamics of slender beams. Josh R. Gladden (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu), Nestor Handzy, Andrew Belmonte (Dept. of Mathematics, The Penn State Univ., University Park, PA), and E. Villermaux (Institut de Recherche sur les Phenomenes Hors Equilibre, Universite de Provence, Marseille, France)

We present experiments on the dynamic buckling of slender rods axially impacted by a projectile. By combining the results of Saint-Venant and elastic beam theory, we derive a preferred wavelength for the buckling instability, and experimentally verify the resulting scaling law for a range of materials using high speed video analysis. The scaling law for the preferred excitation mode depends on the ratio of the longitudinal speed of sound in the beam to the impact speed of the projectile. We will briefly present the imprint of this deterministic mechanism on the fragmentation statistics for brittle beams.

WEDNESDAY MORNING, 29 OCTOBER 2014 MARRIOTT 1/2, 8:00 A.M. TO 9:20 A.M.

Session 3aSAa

Structural Acoustics and Vibration, Architectural Acoustics, and Noise: Vibration Reduction in Air-Handling Systems

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

Chair's Introduction—8:00

Invited Papers

8:05

3aSAa1. Vibration reduction in air handling systems. Angelo J. Campanella (Acculab, Campanella Assoc., 3201 Ridgewood Dr., Ohio, Hilliard, OH 43026, a.campanella@att.net)

Air handling units (AHU) mounted on elevated floors in old and new buildings can create floor vibrations that transmit through the building structure to perturb nearby occupants and sensitive equipment such as electron microscopes. Vibration sources include rotating fan imbalance and air turbulence. Isolation springs and the deflecting floor then create a two degree of freedom system. The analysis discussed here was originally published in "Sound and Vibration," October 1987, pp. 26–30. Analysis parameters will be discussed along with inertia block affects and spring design strategy for floors of finite mass.

8:25

3aSAa2. Determining fan generated dynamic forces for use in predicting and controlling vibration and structure-borne noise from air handling equipment. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

Vibration measurements were conducted to determine the dynamic forces imparted by an operating fan to the floor of an existing rooftop mechanical room. The calculated forces where then used as inputs to a Finite Element Analysis (FEA) computer model to predict the vibration and structure-borne noise in a future building with a similar fan. This paper summarizes the vibration measurements, analysis of the measured data, the subsequent FEA analysis of the future building and the recommendations developed to control fan generated noise and vibration in the future building.

8:45

3aSAa3. Vibration isolation of mechanical equipment: Case studies from light weight offices to casinos. Steve Pettyjohn (The Acoust. & Vib. Group, Inc., 5765 9th Ave., Sacramento, CA CA, spettyjohn@acousticsandvibration.com)

Whether to vibration isolate HVAC equipment or not is often left to the discretion of the mechanical engineer or the equipment supplier. Leaving he isolators out saves money in materials and installation. The value of putting them is not so clear. The cost of not installing the isolators is seldom understood nor the cost of installing them later and the loss of trust by the client. Vibration is generated by all rotating and reciprocating equipment. The resulting unbalanced forces ares seldom known with certainty nor are they quantified. This paper explores the isolation of HVAC equipment on roof, penthouses and roofs without consideration for the stiffness of the structures or resonances of other building elements. The influence of horizontal forces and the installation of the equipment to account for these forces is seldom considered. The application of restraining forces must consider where the force is applied and what the moment arm is. A quick review of the basic formulas will be given one-degree and multi-degree systems. Examples of problems that arose when the vibration isolated was not considered will be presented for a variety of conditions. The corrective actions will be given also.

Contributed Paper

9:05

3aSAa4. Transition of steady air flow into an anharmonic acoustic pulsed flow in a prototype reactor column: Experimental results and mathematical modeling. Hasson M. Tavossi (Phys., Astronomy, & Geo-Sci., Valdosta State University, 2402 Spring Valley Cir, Valdosta, GA 31602, htavossi@valdosa.edu)

A prototype experimental setup is designed to convert steady air flow into an oscillatory anharmonic acoustic pulsed flow, under special experimental conditions. The steady flow in a cylindrical reactor column of 3 m height and 15 cm in diameter with a porous layer, transforms itself abruptly into an oscillatory acoustic pulsed flow. Experimental results show the existence of a threshold for flow-rate, beyond which this transformation into anharmonic oscillatory flow takes place. This change in flow regime is analogous to the phenomenon of bifurcation in a chaotic system, with abrupt change from one energy state into another. Experimental results show that the acoustic oscillations amplitude depends on system size. Preliminary mathematical model will be presented that includes; relaxation oscillations, non-equilibrium thermodynamics, and Joule-Thomson effect. The frequencies at peak amplitude for the acoustic vibrations in the reactor column are expressed in terms of flow-rate, pressure-drop, viscosity, and dimensionless characteristic numbers of the air flow in the system.

WEDNESDAY MORNING, 29 OCTOBER 2014 MARRIOTT 1/2, 10:00 A.M. TO 12:00 NOON

Session 3aSAb

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

10:00

3aSAb1. Design of an experiment to measure unsteady shear stress and wall pressure transmitted through an elastomer in a turbulent boundary layer. Cory J. Smith (Appl. Res. Lab., The Penn State Univ., 1109 Houserville Rd., State College, PA 16801, coryjonsmith@gmail.com), Dean E. Capone, and Timothy A. Brungart (Graduate Program in Acoust., Appl. Res. Lab., The Penn State Univ., State College, PA)

A flat plate that is exposed to a turbulent boundary layer (TBL) experiences unsteady velocity fluctuations which result in fluctuating wall pressures and shear stresses on the surface of the plate. There is an interest in understanding how fluctuating shear stresses and normal pressures generated on the surface of an elastomer layer exposed to a TBL in water are transmitted through the layer onto a rigid backing plate. Analytic models exist which predict these shear stress and normal pressure spectra on the surface of the elastomer as well as those transmitted through the elastomer. The design of a novel experiment is proposed which will utilize Surface Stress Sensitive Films (S3F) to measure the fluctuating shear stress and hydrophones to measure fluctuating normal pressure at the elastomer-plate interface. These experimental measurements would then be compared to

models of unsteady shear and unsteady pressure spectra within a TBL for purposes of model validation. This work will present the design of an experiment to measure the unsteady pressure and unsteady shear at the elastomer-plate interface and the methodology for comparing the measured results to the analytic model predictions.

10:15

3aSAb2. Exploration into the sources of error in the two-microphone transfer function impedance tube method. Hubert S. Hall (Naval Surface Warfare Ctr. Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, hubert.hall@navy.mil), Joseph Vignola, John Judge (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC), and Diego Turo (Dept. of Biomedical Eng., George Mason Univ., Fairfax, VA)

The two-microphone transfer function method has become the most widely used method of impedance tube testing. Due to its measurement speed and ease of implementation, it has surpassed the standing-wave ratio method in popularity despite inherent frequency limitations due to tube geometry. Currently, the two-microphone technique is described in test standards ASTM E1050 and ISO 10534-2 to ensure accurate measurement. However, while detailed for correct test execution, the standards contain vague recommendations for a variety of measurement parameters. For instance, it is only stated in ASTM E1050 that "tube construction shall be massive so sound transmission through the tube wall is negligible." To quantify this requirement, damping of the tube was varied to determine how different loss factor values effect measured absorption coefficient values. Additional sources of error explored are the amount of required absorbing material within the tube for reflective material measurements, additional calibration methods needed for test of excessive reflective materials, and alternate methods of combating microphone phase error and tube attenuation.

10:30

3aSAb3. Analysis of the forced response and radiation of a single-dimpled beam with different boundary conditions. Kyle R. Myers and Koorosh Naghshineh (Mech. & Aerosp. Eng., Western Michigan Univ., College of Eng. & Appl. Sci., 4601 Campus Dr., Kalamazoo, MI 49008, kyle.r.myers@wmich.edu)

Beading and dimpling via the stamping process has been used for decades to stiffen structures (e.g., beams, plates, and shells) against static loads and buckling. Recently, this structural modification technique has been used as a means to shift a structure's natural frequencies and to reduce its radiated sound power. Most studies to date have modeled dimpled beams and dimpled/beaded plates using the finite element method. In this research, an analytical model is developed for a beam with any number of dimples using Hamilton's Principle. First, the natural frequencies and mode shapes are predicted for a dimpled beam in free transverse vibration. A comparison with those obtained using the finite element method shows excellent agreement. Second, the forced response of a dimpled beam is calculated for a given input force. Mode shapes properly scaled from the forced response are used in order to calculate the beam strain energy, thus demonstrating the effect of dimpling on beam natural frequencies. Finally, some preliminary results are presented on the changes in the radiation properties of dimpled beams.

10:45

3aSAb4. The impact of CrossFit training—Weight drops on floating floors. Richard S. Sherren (Kinetics Noise Control, 6300 Irelan Pl., Dublin, OH 43062, rsherren@kineticsnoise.com)

CrossFit training is a popular fitness training method. Some facilities install lightweight plywood floating floor systems as a quick, inexpensive method to mitigate impact generated noise and vibration into adjoining spaces. Part of the CrossFit training regimen involves lifting significant weight overhead, and then dropping the weight on the floor. The energy transferred to the floor system can cause severe damage to floor surfaces and structures; and, when using a lightweight floating floor system, even the isolators can be damaged. This paper describes a spreadsheet based analytical model being used to study the effects of such impacts on various floor systems. This study is a prelude to experiments that will be performed on a full scale model test floor. The results of those experiments will be used to verify the model so that it can be used as a design tool for recommending solutions for mitigating the noise and vibration in adjoining spaces due to floor impact problems. Also discussed in this paper are the qualitative results of some preliminary tests performed in order to better understand the mechanics of impacts on floating floor assemblies.

11:00

3aSAb5. Stethoscope-based detection of detorqued bolts using impact-induced acoustic emissions. Joe Guarino (Mech. and Biomedical Eng., Boise State Univ., Boise, ID) and Robert Hamilton (civil Eng., Boise State Univ., 1910 University Dr., Boise, ID 83725, rhamilton@boisestate.edu)

Non-invasive impact analysis can be used to detect loosened bolts in a steel structure composed of construction-grade I beams. An electronically enhanced stethoscope was used to acquire signals from a moderate to light impact of a hammer on a horizontal steel I beam. Signals were recorded by placing the diaphragm of the stethoscope on the flange of either the horizontal beam or the vertical column proximal to a bolted connection connecting the two members. Data were taken using a simple open-loop method; the input signal was not recorded, nor was it used to reference the output signal. The bolted connection had eight bolts arranged in a standard configuration. Using the "turn of the nut" standard outlined by the Research Council on Structural Connections (RCSC, TDS-012 2-18-08), the bolted joint was tested in three conditions: turn of the nut tight, finger tight, and loose. We acquired time-based data from each of 52 patterns of the eight bolts in three conditions of tightness. Results of both time and frequency-based analyses show that open-loop responses associated with detorqued bolts vary in both amplitude decay and frequency content. We conclude that a basic mechanism can be developed to assess the structural health of bolted joints. Results from this project will provide a framework for further research, including the analysis of welded joints using the same approach.

11:15

3aSAb6. Creep behavior of composite interlayer and its influence on impact sound of floating floor. Tongjun Cho, Byung Kwan Oh, Yousok Kim, and Hyo Seon Park (Architectural Eng., Yonsei Univ., Yonseino 50 Seodaemun-gu, Seoul 120749, South Korea, tjcho@yonsei.ac.kr)

Creep-induced changes in dynamic stiffness of resilient interlayer used for floating floor is an important parameter of vibration isolator in long-term use. Compressive creep behavior of a composite layer made from closed-cell foam and fibrous material is investigated using a Findley equation-based method recommended by International Organization for Standardization (ISO). Quasi-static mechanical analysis is used to evaluate the dynamic stiffness influenced by the creep-deformation of the composite layer. It is shown in the present work that the long-term creep strain of the interlayer under nominal load of the floor and furniture is within the zone where dynamic stiffness increases. The changes in low frequency impact sound by the long-term creep deformation are estimated through real scale laboratory experiments and numerical vibro-acoustic analysis.

11:30

3aSAb7. Investigation of damping in the polymer concrete sleeper for use in reduction of rolling noise from railway. SangKeun Ahn, Eunbeom Jeon, Junhong Park, Hak-sung Kim (Mech. Eng., Hanyang Univ., 222, Wangsimni-ro, Seongdong-gu, Appendix of Eng. Ctr., 211, Seoul 133-791, South Korea, ask9156@hanyang.ac.kr), and Hyo-in Kho (Korea RailRd. Res. Inst., Uiwang, South Korea)

The purpose of this study was to measure damping of various polymer concretes to be used as sleepers for railway. The polymer concretes consisted of epoxy monomer, hardener and aggregates. Various polymer concrete specimens were made by changing epoxy resin weight ratio and curing temperature. The dynamic properties of the polymer concrete specimens were measured by using beam-transfer function method. To predict reduction performance of the polymer concrete sleepers, an infinite Timoshenko beam model was investigated after applying measured concrete properties. The moving loads from rotating wheels on railway due to different roughness were utilized in railway vibration analysis. The vibration response was predicted from which the effects of supporting stiffness and loss factor of sleeper were investigated. The radiated sound power was predicted using calculated rail vibration response. Consequently, the sound power levels

were compared for rails supported by different polymer concrete sleepers. The result of this study assists constructing low noise railway.

11:45

3aSAb8. Study on impulsive noise radiation from of gasoline direct injector. Yunsang Kwak and Junhong Park (Mech. Eng., Hanyang Univ., 515 FTC Hanyang Univ. 222, Wangsimni-ro Seongdong-gu, Seoul ASI|KR|KS013|Seoul, South Korea, toy0511@hanmail.net)

A gasoline direct injection (GDI) engine uses its own injectors for high pressure fuel supply to the combustion chamber. High frequency impact sound during the injection process is one of the main contributors to engine combustion noise. This impact noise is generated during opening and closing by an injector rod operated by a solenoid. For design of an injector with reduced noise generation, it is necessary to analyze its sound radiation mechanism and propose consequent evaluation method. Spectral and modal characteristics of the injectors were measured through vibration induced by external hammer excitation. The injector modal characteristics were analyzed using a simple beam after analyzing its boundaries by complex transverse and rotational springs. To evaluate impulsive sounds more effectively, Prony analysis of sounds was used for verifying influence of injector modal characteristics.

WEDNESDAY MORNING, 29 OCTOBER 2014 — MARRIOTT 5, 8:00 A.M. TO 12:00 NOON

Session 3aSC

Speech Communication: Vowels = Space + Time, and Beyond: A Session in Honor of Diane Kewley-Port

Catherine L. Rogers, Cochair
Dept. of Communication Sciences and Disorders, University of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620

Amy T. Neel, Cochair
Dept. of Speech and Hearing Sci., Univ. of New Mexico, MSC01 1195, University of New Mexico, Albuquerque, NM 87131

Chair's Introduction—8:00

Invited Papers

8:05

3aSC1. Vowels and intelligibility in dysarthric speech. Amy T. Neel (Speech and Hearing Sci., Univ. of New Mexico, MSC01 1195, University of New Mexico, Albuquerque, NM 87131, atneel@unm.edu)

Diane Kewley-Port's work in vowel perception under challenging listening conditions and in the relation between vowel perception and production in second language learners has important implications for disordered speech. Vowel space area has been widely used as an index of articulatory working space in speakers with hypokinetic dysarthria related to Parkinson disease (PD), with the assumption that a larger vowel space is associated with higher speech intelligibility. Although many studies have reported acoustic measures of vowels in Parkinson disease, vowel identification and transcription tasks designed to relate changes in production with changes in perception are rarely performed. This study explores the effect of changes in vowel production by six talkers with PD speaking at habitual and loud levels of effort on listener perception. The relation among vowel acoustic measures (including vowel space area and measures of temporal and spectral distinctiveness), vowel identification scores, speech intelligibility ratings, and sentence transcription accuracy for speakers with dysarthria will be discussed.

8:25

3aSC2. Vowels in clear and conversational speech: Within-talker variability in acoustic characteristics. Sarah H. Ferguson and Lydia R. Rogers (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 was developed for the first author's doctoral dissertation, which was directed by Diane Kewley-Port at Indiana University. While most studies using the Ferguson Database have examined variability among the 41 talkers, the present investigation considered within-talker differences. Specifically, this study examined the amount of variability each talker showed among the 7 tokens of each of 10 vowels produced in clear versus conversational speech. Steady-state formant frequencies have been measured for 5740 vowels in /bVd/ context using PRAAT, and a variety of measures of spread will be used to determine variability for each vowel in each speaking style for each talker. Results will be compared to those of the only known previous study that included a sufficiently large number of tokens for this type of analysis, an unpublished thesis from 1980. Based on this study, we predict that within-token variability will be smaller in clear speech than in conversational speech.

8:45

3aSC3. Understanding speech from partial information: The contributions of consonants and vowels. Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1621 Greene St., Columbia, SC 29208, fogerty@sc.edu)

In natural listening environments, speech is commonly interrupted by background noise. These environments require the listener to extract meaningful speech cues from the partially preserved acoustic signal. A number of studies have now investigated the relative contribution of preserved consonant and vowel segments to speech intelligibility using an interrupted speech paradigm that selectively preserves these segments. Results have demonstrated that preservation of vowel segments results in greater intelligibility for sentences compared to consonant segments, especially after controlling for preserved duration. This important contribution from vowels is specific to sentence contexts and appears to result from suprasegmental acoustic cues. Converging evidence from acoustic and behavioral investigations suggests that these cues are primarily conveyed through temporal amplitude modulation of vocalic energy. Additional empirical evidence suggests that these temporal cues of vowels, conveying the rhythm and stress of speech, are important for interpreting global linguistic cues about the sentence, such as involved in syntactic processing. In contrast, consonant contributions appear to be specific to lexical access regardless of the linguistic context. Work testing older adults with normal and impaired hearing demonstrates their preserved sensitivity to contextual cues conveyed by vowels, but not consonants. [Work supported by NIH.]

9:05

3aSC4. Vowel intelligibility and the second-language learner. Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

Diane Kewley-Port's work has contributed to our understanding of vowel perception and production in a wide variety of ways, from mapping the discriminability of vowel formants in conditions of minimal uncertainty to vowel processing in challenging conditions, such as increased presentation rate and noise. From the results of these studies, we have learned much about the limits of vowel perception for normal-hearing listeners and the robustness of vowels in speech perception. Continuously intertwined with this basic research has been its application to our understanding of vowel perception and vowel acoustics across various challenges, such as hearing impairment and second-language learning. Diane's work on vowel perception and production by second-language learners and ongoing research stemming from her influence will be considered in light of several factors affecting communicative success and challenge for second-language learners. In particular, we will compare the influence of speaking style, noise, and syllable disruption on the intelligibility of vowels perceived and produced by native and non-native English-speaking listeners.

9:25

3aSC5. Vowel formant discrimination: Effects of listeners' hearing status and language background. Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, changliu@utexas.edu)

The goal of this study was to examine effects of listeners' hearing status (e.g., normal and impaired hearing) and language background (e.g., native and non-native) on vowel formant discrimination. Thresholds of formant discrimination were measured for F1 and F2 of English vowels at 70 dB SPL for normal- (NH) and impaired-hearing (HI) listeners using a three-interval, two-alternative forced-choice procedure with a two-down, one-up tracking algorithm. Formant thresholds of HI listeners were comparable to those of NH listeners for F1, but significantly higher than NH listeners for F2. Results of a further experiment indicated that an amplification of the F2 peak could markedly improve formant discrimination for HI listeners, but a simple amplification of the sound level did not provide any benefit to them. On the other hand, another experiment showed that vowel density of listeners' native language appeared to affect vowel formant discrimination, i.e., more crowded vowel space of listeners' native language, better their vowel formant discrimination. For example, English-native listeners showed significantly lower thresholds of formant discrimination for both English and Chinese vowels than Chinese-native listeners. However, the two groups of listeners had similar psychophysical capacity to discriminate formant frequency changes in non-speech sounds.

9:45

3aSC6. Consonant recognition in noise for bilingual children with simulated hearing loss. Kanae Nishi, Andrea C. Trevino (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, kanae.nishi@boystown.org), Lydia Rosado Rogers (Commun. Sci. and Disord., Univ. of Utah, Omaha, Nebraska), Paula B. Garcia, and Stephen T. Neely (Boys Town National Res. Hospital, Omaha, NE)

The negative impacts of noisy listening environments and hearing loss on speech communication are known to be greater for children and non-native speakers than adult native speakers. Naturally, the synergistic influence of listening environment and hearing loss is expected to be greater for bilingual children than their monolingual or normal-hearing peers, but limited studies have explored this issue. The present study compared the consonant recognition performance of highly fluent school-age Spanish-English bilingual children to that of monolingual English-speaking peers. Stimulus materials were 13 English consonants embedded in three symmetrical vowel-consonant-vowel (VCV) syllables. To control for variability in hearing loss profiles, mild-to-moderate sloping sensorineural hearing loss modeled after Pittman & Stelmachowicz [Ear Hear 24, 198–205 (2003)] was simulated following the method used by Desloge *et al.* [Trends Amplification 16(1), 19–39 (2012)]. Listeners heard VCVs in quiet and in the background of speech-shaped noise with and without simulated hearing loss. Overall performance and the recognition of individual consonants will be discussed in terms of the influence of language background (bilingual vs. monolingual), listening condition, simulated hearing loss, and vowel context. [Work supported by NIH.].

10:05–10:20 Break

10:20

3aSC7. Distributions of confusions for the 109 syllable constituents that make up the majority of spoken English. James D. Miller, Charles S. Watson, and Roy Sillings (Res., Commun. Disord. Technol., Inc., 3100 John Hinkle Pl, Ste. 107, Bloomington, IN 47408, jamdmill@indiana.edu)

Among the interests of Kewley-Port have been the perception and production of English Speech Sounds by native speakers of other languages. ESL students from four language backgrounds (Arabic, Chinese, Korean, and Spanish) were enrolled in a speech perception training program. Similarities and differences between these L1 groups in their primary confusions were determined for onsets, nuclei and codas utilized in spoken English. An analysis in terms of syllable constituents is more meaningful than analyses in terms of phonemes as individual phonemes have differing articulatory and acoustic structures depending on their roles in the syllable and their phonetic environments. An important observation is that only a few of all the possible confusions that might occur do occur. Another interesting characteristic of confusions among syllable constituents is that many more confusions are observed than those popularly cited, e.g., the /r/ v /l/ for Japanese speakers. As noted by many, the perceptual problems encountered by learners of English are conditioned on the relations between the sound-structures of English with each talker's L1. These data suggest that the intrinsic similarities within of the sounds of English also play an important role.

10:40

3aSC8. Identification and response latencies for Mandarin-accented isolated words in quiet and in noise. Jonathan Dalby (Commun. Sci. and Disord., Indiana-Purdue, Fort Wayne, 2101 East Coliseum Blvd., Fort Wayne, IN 46805, dalbyj@ipfw.edu), Teresa Barcenas (Speech and Hearing Sci., Portland State Univ., Portland, OR), and Tanya August (Speech-Lang. Pathol., G-K-B Community School District, Garrett, IN)

This study compared the intelligibility of native and foreign-accented American English speech presented in quiet and mixed with two different levels of background noise. Two native American English speakers and two native Mandarin Chinese speakers for whom English is a second language read three 50-word lists of phonetically balanced words (Stuart, 2004). The words were mixed with noise at three different signal-to-noise levels—no noise (quiet), SNR + 10 dB (signal 10 dB louder than noise) and SNR 0 (signal and noise at equal loudness). These stimuli were presented to ten native American English listeners who were simply asked to repeat the words they heard the speakers say. Listener response latencies were measured. The results showed that for both native and accented speech, response latencies increased as the noise level increased. For words identified correctly, response times to accented speech were longer than for native speech but the noise conditions affected both types equally. For words judged incorrectly, however, the noise conditions increased latencies for accented speech more than for native speech. Overall, these results support the notion that processing accented speech requires more cognitive effort than processing native speech.

11:00

3aSC9. The contribution of vowels to auditory-visual speech recognition and the contributions of Diane Kewley-Port to the field of speech communication. Carolyn Richie (Commun. Sci. & Disord., Butler Univ., 4600 Sunset Ave, Indianapolis, IN 46208, crichie@butler.edu)

Throughout her career, Diane Kewley-Port has made enduring contributions to the field of Speech Communication in two ways—through her research on vowels and through her mentoring. Diane has contributed greatly to current knowledge about vowel acoustics, vowel discrimination and identification, and the role of vowels in speech recognition. Within that line of research, Richie & Kewley-Port (2008) investigated the effects of visual cues to vowels on speech recognition. Specifically, we demonstrated that an auditory-visual vowel-identification training program benefited sentence recognition under difficult listening conditions more than consonant-identification training and no training. In this presentation, I will describe my continuing research on the relationship between auditory-visual vowel-identification training and listening effort, for adults with normal hearing. In this study, listening effort was measured in terms of response time and participants were tested on auditory-visual sentence recognition in noise. I will discuss the ways that my current work has been inspired by past research with Diane, and how her mentoring legacy lives on.

11:20

3aSC10. Individual differences in the perception of nonnative speech. Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

As a mentor, Diane Kewley-Port was attentive to each student's needs and took a highly hands-on, individualized approach. In many of her collaborative research endeavors, she has also taken a fine-grained approach toward both discovering individual differences in speech perception and production as well as explaining the causes and consequences of this range of variation. I will present research investigating several cognitive-linguistic factors that may contribute to individual differences in the perception of nonnative speech. Recognizing words from nonnative talkers can be particularly difficult when combined with environmental degradation (e.g., background noise) or listener limitations (e.g., child listener). Under these conditions, the range of performance across listeners is substantially wider than observed under more optimal conditions. My work has investigated these issues in monolingual and bilingual adults and children. Results have indicated that age, receptive vocabulary, and phonological awareness are predictive of nonnative word recognition. Factors supporting native word recognition, such as phonological memory, were less strongly associated with nonnative word recognition. Together, these results suggest that the ability to accurately perceive nonnative speech may rely, at least partially, on different underlying cognitive-linguistic abilities than those recruited for native word recognition. [Work supported by NIH-R21DC010027.]

11:40

3aSC11. Individual differences in sensory and cognitive processing across the adult lifespan. Larry E. Humes (Indiana Univ., Dept. Speech & Hearing Sci., Bloomington, IN 47405-7002, humes@indiana.edu)

A recent large-scale (N = 245) cross-sectional study of threshold sensitivity and temporal processing in hearing, vision and touch for adults ranging in age from 18 through 82 years of age questioned the long-presumed link between aging and declines in cognitive-processing [Humes, L.E., Busey, T.A., Craig, J. & Kewley-Port, D. (2013). Attention, Perception and Psychophysics, 75, 508–524]. The results of this extensive psychophysical investigation suggested that individual differences in sensory processing across multiple tasks and senses drive individual differences in cognitive processing in adults regardless of age. My long-time colleague at IU, Diane Kewley-Port, was instrumental in the design, execution and interpretation of results for this large study, especially for the measures of auditory temporal processing. The methods used and the results obtained in this study will be reviewed, with a special emphasis on the auditory stimuli and tasks involved. The potential implications of these findings, including possible interventions, will also be discussed. Finally, future research designed to better evaluate the direction of the association between sensory-processing and cognitive-processing deficits will be described. [Work supported, in part, by research grant R01 AG008293 from the NIA.]

WEDNESDAY MORNING, 29 OCTOBER 2014 INDIANA G, 8:30 A.M. TO 10:00 A.M.

Session 3aSPa

Signal Processing in Acoustics: Beamforming and Source Tracking

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

Contributed Papers

8:30

3aSPa1. An intuitive look at the unscented Kalman filter. Edmund Sullivan (Res., prometheus, 46 Lawton Brook Ln., Portsmouth, RI 02871, bewegungslos@fastmail.fm)

The Unscented Kalman Filter or UKF is a powerful and easily used modification to the Kalman filter that permits it use in the case of a nonlinear process or measurement model. Its power lies in its ability to allow the mean and covariance of the data to be correctly passed through a nonlinearity, regardless of the form of the nonlinearity. There is a great deal of literature on the UKF that describes the method and gives instruction on its use, but there are no clear descriptions on why it works. In this paper, we show that by computing the mean and covariance as the expectations of a Gaussian process, passing the results through a nonlinearity and solving the resulting integrals using Gauss-Hermite quadrature, the reason for the ability of the UKF to maintain the correct mean and covariance is explained by the fact that the Gauss-Hermite quadrature uses the same abscissas and weights regardless of the form of the integrand.

8:45

3aSPa2. Tracking unmanned aerial vehicles using a tetrahedral microphone array. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil) and R. L. Culver (Appl. Res. Lab., Penn State Univ., State College, PA)

Unmanned Aerial Vehicles (UAVs) present a difficult localization problem for traditional radar systems due to their small radar cross section and relatively slow speeds. To help address this problem, the U.S. Army Research Laboratory (ARL) is developing and testing acoustic-based detection and tracking algorithms for UAVs. The focus has been on detection, bearing and elevation angle estimation using either minimum mean square error or adaptive beamforming methods. A model-based method has been implemented which includes multipath returns, and a Kalman filter has been implemented for tracking. The acoustic data were acquired using ARL's tetrahedral microphone array against several UAV's. While the detection and tracking algorithms perform reasonably well, several challenges remain. For example, interference from other sources resulted in lower signal to interference ratio (SIR), which can significantly degrade performance. The presence of multipath nearly always results in greater variance in elevation angle estimates than in bearing angle estimates.

9:00

3aSPa3. An ultrasonic echo characterization approach based on particle swarm optimization. Adam Pedrycz (Sonic/LWD, Schlumberger, 2-2-1 Fuchinobe, Sagamihara, Kanagawa 229-0006, Japan, APedrycz@slb.com), Henri-Pierre Valero, Hiroshi Hori, Kojiro Nishimiya, Hitoshi Sugiyama, and Yoshino Sakata (Sonic/LWD, Schlumberger, Sagamihara, Kanagawa-ken, Japan)

Presented is a hands-free approach for the extraction and characterization of ultrasonic echoes embedded in noise. By means of model-based nondestructive evaluation approaches, echoes can be represented parametrically by arrival time, amplitude, frequency, etc. Inverting for such parameters is a non-linear task, usually employing gradient-based, least-squared minimization such as Gauss-Newton (GN). To improve inversion stability, suitable initial echo parameter guesses are required which may not be possible under the presence of noise. To mitigate this requirement, particle swarm optimization (PSO) is employed in lieu of GN. PSO is a population-based optimization technique wherein a swarm of particles explores a multidimensional search space of candidate solutions. Particles seek out the global optimum by iteratively moving to improve their position by evaluating their individual performance as well as that of the collective. Since the inversion problem is non-linear, multiple suboptimal solutions exist, and in this regard PSO has a much lower propensity of becoming trapped in a local minima compared to gradient-based approaches. Due to this, it is possible to omit initial guesses and utilize a broad search range instead, which becomes far more trivial. Real pulse-echoes were used to evaluate the efficacy of the PSO approach under varying noise severity. In all cases, PSO characterized the echo correctly while GN required an initial guess within 30% of the true value to converge.

9:15

3aSPa4. Beamspace compressive spatial spectrum estimation on large aperture acoustic arrays. Geoffrey F. Edelmann, Jeffrey S. Rogers, and Steve L. Means (Acoust., Code 7160, U. S. Naval Res. Lab., 4555 Overlook Ave SW, Code 7162, Washington, DC 20375, edelmann@nrl.navy.mil)

For large aperture sonar arrays, the number of acoustic elements can be quite sizable and thus increase the dimensionality of the l1 minimization required for compressive beamforming. This leads to high computational complexity that scales by the cube of the number of array elements. Furthermore, in many applications, raw sensor outputs are often not available since computation of the beamformer power is a common initial processing step performed to reduce subsequent computational and storage requirements. In this paper, a beamspace algorithm is presented that computes the compressive spatial spectrum from conventional beamformer output power. Results from CALOPS-07 experiment will be presented and shown to significantly reduce the computational load as well as increase robustness when detecting low SNR targets. [This work was supported by ONR.]

9:30

3aSPa5. Experimental investigations on coprime microphone arrays for direction-of-arrival estimation. Dane R. Bush, Ning Xiang (Architectural Acoust., Rensselaer Polytechnic Inst., 2609 15th St., Troy, NY 12180, danebush@gmail.com), and Jason E. Summers (Appl. Res. in Acoust. LLC (ARiA), Washington, DC)

Linear microphone arrays are powerful tools for determining the direction of a sound source. Traditionally, uniform linear arrays (ULA) have inter-element spacing of half of the wavelength in question. This produces the narrowest possible beam without introducing grating lobes—a form of aliasing governed by the spatial Nyquist theorem. Grating lobes are often undesirable because they make direction of arrival indistinguishable among their passband angles. Exploiting coprime number theory, however, an array can be arranged sparsely with fewer total elements, exceeding the aforementioned spatial sampling limit separation. Two sparse ULA sub-arrays with coprime number of elements, when nested properly, each produce narrow grating lobes that overlap with one another exactly in just one direction. By combining the sub-array outputs it is possible to retain the shared beam while mostly canceling the other superfluous grating lobes. This work implements two coprime microphone arrays with different lengths and sub-array spacings. Experimental beam patterns are shown to correspond with simulated results even at frequencies above and below the array's design frequency. Side lobes in the directional pattern are inversely correlated with bandwidth of analyzed signals.

9:45

3aSPa6. Shallow-water waveguide invariant parameter estimation and source ranging using narrowband signals. Andrew Harms (Elec. and Comput. Eng., Duke Univ., 129 Hudson Hall, Durham, NC 27708, andrew.harms@duke.edu), Jonathan Odom (Georgia Tech Res. Inst., Durham, North Carolina), and Jeffrey Krolik (Elec. and Comput. Eng., Duke Univ., Durham, NC)

This paper concerns waveguide invariant parameter estimation using narrowband underwater acoustic signals from multiple sources at known range, or alternatively, the ranges of multiple sources assuming known waveguide invariant parameters. Previously, the waveguide invariant has been applied to estimate the range or bottom properties from intensity striations observed from a single broadband signal. The difficulty in separating striations from multiple broadband sources, however, motivates the use of narrowband components, which generally have higher signal-to-noise ratios and are non-overlapping in frequency. In this paper, intensity fluctuations of narrowband components are shown to be related across frequency by a time-warping (i.e., stretching or contracting) of the intensity profile, assuming constant radial source velocity and the waveguide invariant β. A maximum likelihood estimator for the range with β known or for the invariant parameter β with known source range is derived, as well as Cramer-Rao bounds on estimation accuracy assuming a Gaussian noise model. Simulations demonstrate algorithm performance for constant radial velocity sources in a representative shallow-water ocean waveguide. [Work supported by ONR.]

WEDNESDAY MORNING, 29 OCTOBER 2014 — INDIANA G, 10:15 A.M. TO 12:00 NOON

Session 3aSPb

Signal Processing in Acoustics: Spectral Analysis, Source Tracking, and System Identification (Poster Session)

R. Lee Culver, Chair

ARL, Penn State University, PO Box 30, State College, PA 16804

All posters will be on display from 10:15 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 10:15 a.m. to 11:00 a.m. and contributors of even-numbered papers will be at their posters from 11:00 a.m. to 12:00 noon.

Contributed Papers

3aSPb1. Improvement of the histogram in the degenerate unmixing estimation technique algorithm. Junpei Mukae, Yoshihisa Ishida, and Takahiro Murakami (Dept. of Electronics and Bioinformatics, Meiji Univ., 1-1-1 Higashi-mita, Tama-ku, Kawasaki-shi 214-8571, Japan, ce41094@meiji.ac.jp)

A method of improving the histogram in the degenerate unmixing estimation technique (DUET) algorithm is proposed. The DUET algorithm is one of the methods of blind signal separation (BSS). The BSS framework is to retrieve source signals from mixtures of them without a priori information about the source signals and the mixing process. In the DUET algorithm, the histogram of both the direction-of-arrivals (DOAs) and the distances is formed from the mixtures which are observed using two sensors. And then, signal separation is achieved using time-frequency masking based on the histogram. Consequently, the capability for the DUET algorithm strongly depends on the performance of the histogram. In general, the histogram is degraded by the reverberation or the reflection of the source signals when the DUET algorithm is applied in the real environment. Our approach is to

improve the histogram of the DUET algorithm. In our method, the phase component of the observed mixture at each frequency bin is modified by those at the neighboring frequency bins. The proposed method gives the sharper histogram in comparison with the conventional approach.

3aSPb2. Start point estimation of a signal in a frame. Anri Ota (Dept. of Electronics and Bioinformatics, Meiji Univ., 1-1-1 Higashi-mita, Tama-ku, Kawasaki-shi 214-8571, Japan, ce41017@meiji.ac.jp), Yoshihisa Ishida, and Takahiro Murakami (Dept. of Electronics and Bioinformatics, Meiji Univ., Kawashiki-shi, Japan)

An algorithm for start-point estimation of a signal from a frame is presented. In many applications of speech signal processing, the signal to be processed is often segmented into several frames, and then the frames are categorized into speech and non-speech frames. Instead, we focus on only the frame in which the speech starts. To simplify the problem, we assume that the speech is modeled by a number of complex sinusoidal signals. When a complex sinusoidal signal that starts in a frame is observed, it can be modeled as multiplication of a complex sinusoidal signal of which the length is infinite and a window function that has finite duration in the time domain. In the frequency domain, the spectrum of the signal of the frame is given by the shifted spectrum of the window function. Sharpness of the spectrum of the window function depends on the start point of the signal. Hence, the start point of the signal is estimated by the sharpness of the observed spectrum. This approach can be extended to the signal that consists of a number of complex sinusoidal signals. Simulation results using artificially generated signals show the validity of our method.

3aSPb3. Examination and development of numerical methods and algorithms designed for the determination of an enclosure's acoustical characteristics via the Schroeder Function. Miles Possing (Acoust., Columbia College Chicago, 1260 N Dearborn, 904, Chicago, IL 60610, miles@possing.com)

A case study was conducted to measure the acoustical properties of a church auditorium. While modeling the project using EASE 2.1, some problems arose when attempting to determine the reverberation time using the Schroder Back Integrated Impulse Function within EASE 2.1. An auxiliary investigation was launched aiming to better understand the Schroeder algorithm in order to produce a potentially improved version in MATLAB. It was then theorized that the use of a single linear regression is not sufficient to understand the nature of the decay, due to the non-linearity of the curve, particularly during the initial decay. Rather, it is hypothesized that the use of numerical methods to find instantaneous rates of change over the entire initial decay along with a Savitsky-Golay Filter could possibly yield much more robust, accurate results when attempting to derive the local reverberation time from reflectogram data.

3aSPb4. A modified direction-of-arrival estimation algorithm for acoustic vector sensors based on Unitary Root-MUSIC. Junyuan Shen, Wei Li, Yuanming Guo, and Yongjue Chen (Electronics and Commun. Eng., Harbin Inst. of Technol., XiLi University Town HIT C#101, Shenzhen, GuangDong GD 755, China, Juny_Shen@hotmail.com)

A novel method applying for direction-of-arrival(DOA) using acoustic vector sensors(AVS) based on Unitary Root-MUSIC algorithm(URM) is proposed in this paper. AVS array has a characteristic named coherence principle of sound pressure and velocity, which can significantly improve the detection performance of DOA by reducing the influence of white Gaussian noise. We apply this characteristic and the extra velocity information of AVS to construct a modified covariance matrix. In particular, the modified covariance matrix need not extend the dimension in calculation of AVS covariance matrix which means saving the computing time. In addition, we combine the characteristics of modified matrix with URM algorithm to design a new algorithm, which can minimize the impact of environment noise and further reduce computational complexity to a lower order of magnitude. So the proposed method can not only improve the accuracy of DOA detection but also reduce the computational complexity, compared to the classic DOA algorithm. Theory analysis and simulation experiment show that the proposed algorithm for AVS based on URM can significantly improve the DOA resolution in low SNR ratios and few snapshots.

3aSPb5. Multiple pitch estimation using comb filters considering overlap of frequency components. Kyohei Tabata, Ryo Tanaka, Hiroki Tanji, Takahiro Murakami, and Yoshihisa Ishida (Dept. of Electronics and Bioinformatics, Meiji Univ., 1-1-1 Higashimita, Tama-ku, Kawasaki-shi, Kanagawa 214-8571, Japan, ce31063@meiji.ac.jp)

We propose a method of the multiple pitch estimation using the comb filters for transcript. We can know the pitches of a musical sound by detecting the bigger outputs in comb filters connected in parallel. Each comb filter has peak corresponding to each pitch and its harmonic frequencies. The outputs of the comb filters corresponding to input pitch frequencies have bigger frequency components, and show bigger outputs than other comb filter's ones. However, when there is the fundamental frequency of higher tone near harmonics of lower tones, the pitch estimation often fails. Therefore, the estimation is assigned to a wrong note when frequency components are shared. The proposed method estimates the correct pitch by correcting the outputs using the matrix, which is defined as the power ratio of the harmonic frequencies to the fundamental frequency. The effectiveness of our proposed method is confirmed by simulations. The proposed method enables more accurate pitch estimation than other conventional methods.

3aSPb6. Evaluating microphones and microphone placement for signal processing and automatic speech recognition of teacher-student dialog. Michael C. Brady, Sydney D'Mello, Nathan Blanchard (Comput. Sci., Univ. of Notre Dame, Fitzpatrick Hall, South Bend, IN 46616, mbrady8@nd.edu), Andrew Olney (Psych., Univ. of Memphis, Memphis, TN), and Martin Nystrand (Education, English, Univ. of Wisconsin, Madison, WI)

We evaluate a variety of audio recording techniques for a project on the automatic analysis of speech dialog in middle school and high school classrooms. In our scenario, the teacher wears a headset microphone or a lapel microphone. A second microphone is then used to collect speech and related sounds from students in the classroom. Various boundary microphones, omni-directional microphones, and cardioid microphones are tested as this second classroom microphone. A commercial microphone array [Microsoft Xbox Kinect] is also tested. We report on how well digital source-separation techniques work for segregating the teacher and student speech signals from one another based on these various microphones and placements. We also test the recordings using various automatic speech recognition engines for word recognition error rates under different levels of background noise. Preliminary results indicate one boundary microphone, the Crown PZM-30, to be superior for the classroom recordings. This is based on its performance at capturing near and distant student signals for ASR in noisy conditions, as measured by ASR error rates across different ASR engines.

Session 3aUW

Underwater Acoustics, Acoustical Oceanography, Animal Bioacoustics, and ASA Committee on Standards: Standardization of Measurement, Modeling, and Terminology of Underwater Sound

Susan B. Blaeser, Cochair
Acoustical Society of America Standards Secretariat, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747

Michael A. Ainslie, Cochair
Underwater Tech. Dept., TNO, P.O. Box 96864, The Hague 2509JG, Netherlands

George V. Frisk, Cochair
Dept. of Ocean Eng., Florida Atlantic Univ., Dania Beach, FL 33004-3023

Chair's Introduction—9:00

Invited Papers

9:05

3aUW1. Strawman outline for a standard on the use of passive acoustic towed arrays for marine mammal monitoring and mitigation. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu)

There is a perceived need from several U.S. federal agencies and departments to develop consistent standards for how passive acoustic monitoring (PAM) for marine mammals is implemented for mitigation and regulatory monitoring purposes. The use of towed array technology is already being required for geophysical exploration activities in the Atlantic Ocean and the Gulf of Mexico. However, to date no specific standards have been developed or implemented for towed arrays. Here, a strawman outline for a ANSI standard is presented (http://wp.me/P4j34t-a) to cover requirements and recommendations for the following aspects of towed array operations: initial planning (including guidelines for when PAM is not appropriate), hardware, software, and operator training requirements, real-time mitigation and monitoring procedures, and required steps for performance validation. The outline scope, at present, does not cover operational shutdown decision criteria, sound source verification, or defining the required detection range of the system. Instead of specifying details of towed array systems, the current strategy is to focus on the process of defining the required system performance for a given application, and then stepping through how the system hardware, software, and operations should be selected and validated to meet or exceed these requirements. [Work supported by BSEE.]

9:30

3aUW2. Towards a standard for the measurement of underwater noise from impact pile driving in shallow water. Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng. Dept., Univ. of Washington, Mech. Eng., 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu), Pete D. Theobald, and Stephen P. Robinson (National Physical Lab., Children's Respiratory and Critical Care Specialists, PA, Middlesex, United Kingdom)

Measurements of the underwater noise field from impact pile driving are essential to the address environmental regulations in effect in both Europe and North America to protect marine life. For impact pile driving in shallow water there exists a range scale $R^* = H/\tan(\Theta)$ that delineates important features in the propagation of underwater sound from impact pile driving, where Θ is the Mach angle of the wavefront radiated into the water from the pile and H is water depth. This angle is about 17o for many steel piles typically used, and thus R^* is approximately 3H. For range R, such that $R/R^* \sim 0.5$ depth variation in the noise field is highest, more so with peak pressure than with sound exposure level (SEL); for $R/R^* > 1$ the field becomes more uniform with depth. This effect of measurement range can thus have implications on environmental monitoring designed to obtain a close-range datum, which is often used with a transmission loss model to infer the noise level at farther range. More consistent results are likely obtained if the measurement range is at least 3H. Ongoing standardization activities for the measurement and reporting of sound levels radiated from impact pile driving will also be discussed.

9:55

3aUW3. Importance of metrics standardization involving the effects of sound on fish. Michele B. Halvorsen (CSA Ocean Sci. Inc, 8502 SW Kansas Hwy, Stuart, FL 34997, mhalvorsen@conshelf.com)

Reporting accurate metrics while employing good measurement practices is a topic that is gaining awareness. Seemingly a simple and expected task, however when reading current and past literature, reporting sound metrics utilized is often not met. It is clear that increased awareness and development of standardization of acoustic metrics is necessary. When reviewing previously published literature on the effects of sound on fish, it is often difficult to fully understand how metrics were calculated leaving the reader to make assumptions. Furthermore, the lack of standardization and definition decreases the amount of data and research studies that could be directly comparable. In a field that has paucity of effects of sound on fish, this situation underscores the importance and need for standardization.

10:20

3aUW4. Developments in standards and calibration methods for hydrophones and electroacoustic transducers for underwater acoustics. Stephen P. Robinson (National Physical Lab., National Physical Lab., Hampton Rd., Teddington TW11 0LW, United Kingdom, stephen.robinson@npl.co.uk), Kenneth G. Foote (Woods Hole Oceanographic Inst., Woods Hole, MA), and Pete D. Theobald (National Physical Lab., Teddington, United Kingdom)

If they are to be meaningful, underwater acoustic measurements must be related to common standards of measurement. In this paper, a description is given of the existing standards for the calibration of hydrophones and electroacoustic transducers for underwater acoustics. The description covers how primary standards are currently realized and disseminated, and how they are validated by international comparisons. A report is also given of the status of recent developments in specification standards, for example within the International Organization for Standardization (ISO) and the International Electrotechnical Commission (IEC). The discussion focuses on the revision of standards for transducer calibration, and the inclusion of extended guidance on uncertainty assessment, and on the criteria for determining the locations of the acoustic near-field and far-field. A description is then provided of recent developments using non-traditional techniques such as optical sensing, which may lead to the next generation of standards. A report is also given of the status of recent developments in and of a number of current initiatives to promote best measurement practice.

Contributed Papers

10:45

3aUW5. All clicks are not created equally: Variations in high-frequency acoustic signal parameters of the Amazon river dolphin (*Inia geoffrensis*). Marie Trone (Math and Sci., Valencia College, 1800 Denn John Ln., Kissimee, FL 34744, mtronedolphin@yahoo.com), Randall Balestriero (Université de Toulon, La Garde, France), Hervé Glotin (Université de Toulon, Toulon, France), and Bonnett E. David (None, None, Silverdale, WA)

The quality and quantity of acoustical data available to researchers are rapidly increasing with advances in technology. Recording cetaceans with a 500 kHz sampling rate provides a more complete signal representation than traditional sampling at 96 kHz and lower. Such sampling provides a profusion of data concerning various parameters, such as click duration, interclick intervals, frequency, amplitude, and phase. However, there is disagreement in the literature in the use and definitions of these acoustic terms and parameters. In this study, Amazon River dolphins (*Inia geoffrensis*) were recorded using a 500 kHz sampling rate in the Peruvian Amazon River watershed. Subsequent spectral analyses, including time waveforms, fast Fourier transforms and wavelet scalograms, demonstrate acoustic signals with differing characteristics. These high frequency, broadband signals are compared, and differences are highlighted, despite the fact that currently an unambiguous way to describe these acoustic signals is lacking. The need for standards in cetacean bioacoustics with regard to terminology and collection techniques is emphasized.

11:00

3aUW6. Acoustical terminology in the *Sonar Modelling Handbook*. Andrew Holden (Dstl, Dstl Portsdown West, Fareham PO17 6AD, United Kingdom, apholden@dstl.gov.uk)

The UK *Sonar Modelling Handbook* (SMH) defines the passive and active Sonar Equations, and their individual terms and units, which are extensively used for sonar performance modelling. The new Underwater Acoustical Terminology ISO standard, which is currently being developed by the ISO working group TC43/SC3/WG2 to standardize terminology will have an impact on the SMH definitions. Work will be presented comparing the current SMH terminology with both the future ISO standard and other well-known definitions to highlight the similarities and differences between each of these.

11:15

3aUW7. The definitions of "level," "sound pressure," and "sound pressure level" in the International System of Quantities, and their implications for international standardization in underwater acoustics. Michael A. Ainslie (Acoust. and Sonar, TNO, P.O. Box 96864, The Hague 2509JG, Netherlands, michael.ainslie@tno.nl)

The International System of Quantities (ISQ), incorporating definitions of physical quantities and their units, was completed in 2009 following an extensive collaboration between two major international standards organizations, the International Organization for Standardization (ISO) and the International Electrotechnical Commission (IEC). The ISQ encompasses all SI units as well as selected units outside the SI such as the byte (including both decimal and binary multiples), bel, neper, and decibel. The ISQ, which includes definitions of the terms "level," "sound pressure," and "sound pressure level," is presently being used to underpin an underwater acoustics terminology standard under development by ISO. For this purpose, pertinent ISQ definitions are analyzed and compared with alternative standard definitions, and with conventional use of the same terms. The benefits of combining IEC and ISO definitions into a single standard, solving some longstanding problems, are described. The comparison also reveals some teething problems, such as internal inconsistencies within the ISQ, and discrepancies with everyday use of some of the terms, demonstrating the need for continued collaboration between the major standards bodies. As of 2014, the ISQ is undergoing a major revision, leading to a unique opportunity to resolve these discrepancies.

Session 3pAA

Architectural Acoustics: Architectural Acoustics Medley

Norman H. Philipp, Chair
Geiler & Associates, 1840 E. 153rd Circle, Olathe, KS 66062

Contributed Papers

1:00

3pAA1. From the sound up: Reverse-engineering room shapes from sound signatures. Willem Boning and Alban Bassuet (Acoust., ARUP, 77 Water St., New York, NY 10005, willem.boning@arup.com)

Typically, architects and acousticians design rooms for music starting from a model room shape known from past experience to perform well acoustically. We reverse the typical design process by using a model sound signature to generate room shapes. Our method builds off previous research on reconstructing room shapes from recorded impulse responses, but takes an instrumental, design-oriented approach. We demonstrate how an abstract sound signature constructed in a hybrid image source-statistical acoustical simulator can be translated into a room shape with the aid of a parametric design interface. As a proof of concept, we present a study in which we generated a series of room shapes from the same sound signature, analyzed them with commercially available room acoustic software, and found objective parameters for comparable receiver positions between shapes to be within just-noticeable-difference ranges of each other.

1:15

3pAA2. Achieving acoustical comfort in restaurants. Paul Battaglia (Architecture, Univ. at Buffalo, 31 Rose Ct Apt. 4, Snyder, NY 14226, plb@buffalo.edu)

The achievement of a proper acoustical ambiance for restaurants has long been described as a problem of controlling noise to allow for speech intelligibility among patrons at the same table. This simplification of the acoustical design problem for restaurants does not entirely result in achieving either a sensation of acoustical comfort or a preferred condition for social activity sought by architects. In order to more fully study the subjective impression of acoustical comfort a large data base from 11 restaurants with 75 patron surveys for each (825 total) was assembled for analysis. The results indicate that a specific narrow range of reverberation time can produce acoustical comfort for restaurant patrons of all ages. Other physical and acoustical conditions of the dining space are shown to have little to no consistent effect on the registration of comfort. The results also indicate that different subjective components of acoustical comfort—quietude, communication, privacy—vary significantly by age group with specific consequences for the acoustical design of restaurants for different clienteles.

1:30

3pAA3. 500-seat theater in the city of Qom; Computer simulation vs. acoustics measurements. Hassan Azad (Architecture, Univ. of Florida, 3527 SW, 20th Ave., 1132B, Gainesville, FL 32607, h.azad@ufl.edu)

There is an under construction 500-seat Theater in Qom city in Iran in which I was part of the acoustics design team. We went through a different steps of the acoustics design using Odeon software packages which enabled us to go back and forth in design process and make proper improvement while we were suffering from having limitations on material choice. Fortunately the theater is being built so after a while it would be feasible to do acoustics measurements with the help of Building and Housing Research Center (BHRC) in Iran as well as subjective evaluation during the very first performances. This paper is aimed to juxtapose the results of computer simulation and acoustics measurement and make a comparison in between to see if there are any discrepancies.

1:45

3pAA4. Acoustical materials and sustainability analyses. Hassan Azad (Architecture, Univ. of Florida, 3527 SW, 20th Ave., 1132B, Gainesville, FL 32607, h.azad@ufl.edu)

Acoustical materials can perform a variety of functions from absorption and diffusion to the insulation and noise control. They may have similar acoustical performance but very different characteristics in terms of sustainability. It is important to evaluate the environmental effects of materials which exhibit the same acoustical performance in order to wisely choose the best alternative available. This study is intended to introduce and compare the different tools and methods which are commonly used in environmental sustainability analysis of materials including Eco-profile, Eco-indicator, Eco-invent, and also software packages like IMPACT. Also, a specific kind of computer model is proposed in which one can run process of calculation of both acoustics properties and sustainability assessment of a given material through computer aided techniques. The model consists of a simple cubic room with a given set of materials for its elements like walls, floor, ceiling, and windows or doors (if any). The acoustics properties which can be calculated are reverberation time with the help of either Odeon or Catt-Acoustics Software and Air borne/impact sound insulation with the help of recently developed software, SonArchitect. For the sustainability assessment both LCA method and software packages like IMPACT are the main tools.

2:00

3pAA5. Influence of the workmanship on the airborne sound insulation properties of light weight building plasterboard steel frame wall systems. Herbert Muellner (Acoust. and Bldg. Phys., Federal Inst. of Technol. TGM Vienna, Wexstrasse 19-23, Vienna A-1200, Austria, herbert.muellner@tgm.ac.at) and Thomas Jakits (Appl. Res. and Development, Saint-Gobain Rigips Austria GesmbH, Vienna, Austria)

Building elements which are built according to the light weight mode of construction, e.g. plasterboard steel frame wall systems show a large variation of air borne sound insulation properties although the elements appear as identical. According to several studies conducted in the recent years, certain aspects of workmanship have significant influence on the air borne sound insulation characteristics of light weight building elements. The method to fasten the planking (e.g., gypsum boards, gypsum fiber boards) as well as the number and position of the screws can lead to considerable variations regarding the sound insulation properties. Above 200 Hz, the sound reduction index R can differ more than 10 dB by the variation of the position of the screws. Applying prefabricated composite panels of adhesive connected plasterboards not only considerably reduces the depth of the dip of the critical frequency caused by the higher damping due to the interlayer but it can also significantly decrease the negative influence of the workmanship on the air borne sound insulation properties of these kinds of light weight walls in comparison to the standard planking of double layer plasterboard systems. The influence of secondary construction details and workmanship will be discussed in the paper.

2:15

3pAA6. Contribution of floor treatment characteristics to background noise levels in health care facilities, Part 1. Adam L. Paul, David A. Arena, Eoin A. King, Robert Celmer (Acoust. Prog. & Lab, Univ. of Hartford , 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu), and John J. LoVerde (Paul S. Veneklasen Res. Foundation, Santa Monica, CA)

Acoustical tests were conducted on five types of commercial-grade flooring to assess their potential contribution to noise generated within health care facilities outside of patient rooms. The floor types include sheet vinyl (with and without a 5 mm rubber backing), virgin rubber (with and without a 5 mm rubber backing), and a rubber-backed commercial grade carpet for comparison. The types of acoustical tests conducted were ISO-3741 compliant sound power level testing (using two source types: a tapping machine to simulate footfalls and a rolling hospital cart), and sound absorption testing as per ASTM-C423. Among the non-carpet samples, the material type that produced the least sound power was determined to be the rubber-backed sheet vinyl. While both 5 mm-backed samples showed a significant difference compared to their un-backed counterparts with both source types, the rubber-backed sheet vinyl performed slightly better than the rubber-backed virgin rubber in the higher frequency bands in both tests. The performance and suitability of these flooring materials in a health care facility compared to commercial carpeting will be discussed. [Work supported by Paul S. Veneklasen Research Foundation.]

2:30

3pAA7. Visualization of auditory masking for firefighter alarm detection. Casey Farmer (Dept. of Mech. Eng., Univ. of Texas at Austin, 1208 Enfield Rd., Apt. 203, Austin, TX 78703, caseymfarmer@utexas.edu), Mustafa Z. Abbasi, Preston S. Wilson (Appl. Res. Labs, Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX), and Ofodike A. Ezekoye (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

An essential piece of firefighter equipment is the Personal Alert Safety System (PASS), which emits an alarm when a firefighter has been inactive for a specified period of time and is used to find and rescue downed firefighters. The National Institute for Occupational Safety and Health (NIOSH) firefighter fatality reports suggest that there have been instances when the PASS alarm is not audible by other firefighters on the scene. This paper seeks to use acoustic models to measure the sound pressure level of various signals throughout a structure. With this information, a visual representation will be created to map where a PASS alarm is audible and where it is masked by noise sources. This paper presents an initial audibility study, including temporal masking and frequency analysis. The results will be compared to auralizations and experimental data. Some other potential applications will be briefly explored.

2:45

3pAA8. Investigations on acoustical coupling within single-space monumental structures using a diffusion equation model. Zühre Sü Gül (R&D / Architecture, MEZZO Studyo / METU, METU Technopolis KOSGEB-TEKMER No112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com), Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), and Mehmet Čalışkan (Dept. of Mech. Eng., Middle East Tech. Univ. / MEZZO Studyo, Ankara, Turkey)

Sound energy distributions and flows within single-space rooms can be exploited to understand the occurrence of multi-slope decays. In this work, a real-size monumental worship space is selected for investigations of non-exponential sound energy decays. Previous field tests in this single-space venue indicate multi-slope formation within such a large volume and the multiple-dome upper structure layout. In order to illuminate/reveal the probable reasons of non-exponential sound energy decays within such an architectural venue, sound energy distributions and energy flows are investigated. Due to its computational efficiency and advantages in spatial energy density and flow vector analysis, a diffusion equation model (DEM) is applied for modeling sound field of the monumental worship space. Preliminary studies indicate good agreement for overall energy decay time estimations among experimental field and DEM results. The energy flow vector and energy distribution analysis indicate the upper central dome-structure to be the potential energy accumulation/ concentration zone, contributing to the later energy decays.

WEDNESDAY AFTERNOON, 29 OCTOBER 2014 — INDIANA A/B, 1:00 P.M. TO 3:20 P.M.

Session 3pBA

Biomedical Acoustics: History of High Intensity Focused Ultrasound

Lawrence A. Crum, Cochair
Applied Physics Laboratory, University of Washington, Center for Industrial and Medical Ultrasound, Seattle, WA 98105

Narendra T. Sanghvi, Cochair
R & D, SonaCare Medical, 4000 Pendleton way, Indianapolis, IN 46226

Invited Papers

1:00

3pBA1. History of high intensity focused ultrasound, Bill and Frank Fry and the Bioacoustics Research Laboratory. William O'Brien and Floyd Dunn (Elec. Eng., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, wdo@uiuc.edu)

1946 is a key year in the history of HIFU. That year, sixty-eight years ago, the Bioacoustics Research Laboratory was established at the University of Illinois. Trained in theoretical physics, William J, (Bill) Fry (1918–1968) left his graduate studies at Penn State University to work at the Naval Research Laboratory in Washington, DC on underwater sound during World War II. Bill was hired by the

University of Illinois in 1946, wanting to continue to conduct research activities of his own choosing in the freer university atmosphere. Like Bill, Francis J. (Frank) Fry (1920–2005) went to Penn State as well as the University of Pittsburgh where he studied electrical engineering. Frank joined Bill at the University of Illinois, also in 1946, having worked at Westinghouse Electric Corporation where his division was a prime contractor on the Manhattan Project. Floyd Dunn also arrived at the University of Illinois in 1946 as an undergraduate student, having served in the European Theater during World War II. The talk will recount some of the significant HIFU contributions that emerged from BRL faculty, staff, and students. [NIH Grant R37EB002641.]

1:20

3pBA2. Transforming ultrasound basic research in to clinical systems. Narendra T. Sanghvi and Thomas D. Franklin (R & D, SonaCare Medical, 4000 Pendleton way, Indianapolis, IN 46226, narensanghvi@sonacaremedical.com)

In late 1960s, Robert F. Heimburger, MD, Chief of Neurosurgery at Indiana University School of Medicine, started collaborating with William J. Fry and Francis J. Fry at Interscience Research Institute (IRI) in Champaign, IL. and treated brain cancer patients with HIFU. In 1970, Dr. Heimburger and Indiana University School of Medicine (IUMS) invited IRI to join IUMS and Indianapolis Center For Advanced Research, Inc. (ICFAR). In 1972, a dedicated Fortune Fry Research Laboratory (FFRL) was inaugurated to advance ultrasound research relevant for clinical use. In the '70s, an automated computer controlled, integrated B-mode, image-guided HIFU system ("the candy machine") was developed that successfully treated brain cancer patients at IUMS. HIFU was found to be safe for the destruction of brain tumors. Later a second-generation brain HIFU device was developed to work with CAT or MR images. In 1974, the FFRL developed a first cardiac real-time, 2-D ultrasound scanner. Prof. H. Feigenbaum pioneered this imaging technique and formed "Echocardigraphy Society." In 1978, an automated breast ultrasound system was successfully developed led to form Labsonics, Inc. that proliferated 300 scanners in 4 years. In 1986, the Sonablate system to treat prostate cancer was developed. The Sonablate has been used worldwide.

1:40

3pBA3. The development of high intensity focused ultrasound in Europe, what could we have done better? Gail ter Haar (Phys., Inst. of Cancer Res., Phys. Dept., Royal Marsden Hospital, Sutton, Surrey SM2 5PT, United Kingdom, gail.terhaar@icr.ac.uk)

The clinical uptake of HIFU has been disappointingly slow. This despite its promise as a minimally invasive, ultimately conformal technique. It may be instructive to look at the way in which this technique has evolved from its early days with an eye as to whether a different approach might have resulted in its more rapid acceptance. Examples will be drawn from HIFU's development in the United Kingdom.

2:00

3pBA4. LabTau's experience in therapeutic ultrasound : From lithotripsy to high intensity focused ultrasound. Jean-Yves Chapelon, Michael Canney, David Melodelima, and Cyril Lafon (U1032, INSERM, 151 Cours Albert Thomas, Lyon 69424, France, jean-yves.chapelon@inserm.fr)

Research on therapeutic ultrasound at LabTau (INSERM Lyon, France) began in the early 1980s with work on shock waves that lead to the development of the first ultrasound-guided lithotripter. In 1989, this research shifted towards new developments in the field of HIFU with applications in urology and oncology. The most significant developments have been obtained in urology with the Ablatherm^{TM} project, a transrectal HIFU device for the thermal ablation of the prostate. This technology has since become an effective therapeutic alternative for patients with localized prostate cancer. Since 2000, three generations of the Ablatherm^{TM} have been CE marked and commercialized by EDAP-TMS. The latest version, the FocalOne^{TM}, allows for the focal treatment of prostate cancer and combines dynamic focusing and fusion of MR images to ultrasound images acquired in real time by the imaging probe integrated in the HIFU transducer. Using toroidal ultrasound transducers, a HIFU device was also recently validated clinically for the treatment of liver metastases. Another novel application that has reached the clinic is for the treatment of glaucoma using a miniature, disposable HIFU device. Today, new approaches are also being investigated for treating cerebral and cardiac diseases.

2:20

3pBA5. High intensity therapeutic ultrasound research in the former USSR in the 1950s–1970s. Vera Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., 1013 NE 40th St., Seattle, Washington 98105, va.khokhlova@gmail.com), Valentin Burov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Leonid Gavrilov (Andreev Acoust. Inst., Moscow, Russian Federation)

A historical overview of therapeutic ultrasound research performed in the former USSR in the 1950s–1970s is presented. In the 1950s, the team of A.K.Burov in Moscow proposed the use of non-thermal, non-cavitational mechanisms of high intensity unfocused ultrasound to induce specific immune responses in treating Brown Pearce tumors in an animal model and melanoma tumors in a number of patients. Later, in the early 1970s, new studies began at the Acoustics Institute in Moscow jointly with several medical institutions. Significant results included first measurements of cavitation thresholds in animal brain tissues *in vivo* and demonstration of the feasibility to apply high intensity focused ultrasound (HIFU) for local ablation of brain structures through the intact skull. Another direction was ultrasound stimulation of superficial and deep receptors in humans and animals using short HIFU pulses; these studies became the basis for ultrasound stimulation of different neural structures and have found useful clinical applications for diagnostics of skin, neurological, and hearing disorders. Initial studies on the synergism between ultrasound in therapeutic doses combined with consecutive application of ionizing radiation were carried out. Later, hyperthermia research was also performed for brain tissues and for ophthalmology. [Work supported by the grant RSF 14-12-00974.]

2:40

3pBA6. The development of MRI-guided focused ultrasound at Brigham & Women's Hospital. Nathan McDannold (Radiology, Brigham and Women, 75 Francis St, Boston, MA MA, njm@bwh.harvard.edu)

The Focused Ultrasound Laboratory was created in the Department of Radiology at Brigham & Women's Hospital in the early 1990's, when Ferenc Jolesz invited Kullervo Hynynen to join him to collaborate with GE Medical Systems to develop MRI-guided Focused Ultrasound surgery. This collaboration between Dr. Hynynen, an experienced researcher of therapeutic ultrasound, Dr. Jolesz, who developed MRI-guided laser ablation, and the engineers at GE and later InSightec, with their decades of experience developing MRI and ultrasound systems, established a program that over two decades produced important contributions to HIFU. In this talk, Nathan McDannold, the current director of the laboratory, will review the achievements made by the team of researchers, which include the development of the first MRI-guided FUS system, the creation of the first MRI-compatible phased arrays, important contributions to the validation and implantation of MR temperature mapping and thermal dosimetry, the development of an MRI-guided transcranial system, and the discovery that ultrasound and microbubbles can temporarily disrupt the blood–brain barrier. This output of this team, which led to clinical systems that have treated tens of thousands of patients at sites around the world, is an excellent example of how academic research can be to the clinic.

3:00

3pBA7. What have we learned about shock wave lithotripsy in the past thirty years? Pei Zhong (Mech. Eng. and Mater. Sci., Duke Univ., 101 Sci. Dr., Durham, NC 27708, pzhong@duke.edu)

Shock wave lithotripsy (SWL) has revolutionized the treatment of kidney stone disease since its introduction in the early 1980s. Considering the paucity of knowledge about the bioeffects of shockwaves in various tissues and renal concretions 30 years ago, the success of SWL is a truly remarkable fest on its own. We have learned a lot since then. New technologies have been introduced for shock wave generation, focusing, and measurement, among others. In parallel, new knowledge has been acquired progressively about the mechanisms of stone comminution and tissue injury. Yet there are still outstanding issues that are constantly debated, waiting for resolution. In this talk, the quest for a better understanding of the shockwave interaction with stones and renal tissue in the field of SWL will be reviewed in chronological order. Focus will be on stress waves and cavitation for their distinctly different (for their origin), yet often synergistically combined (in their action), roles in the critical processes of SWL. This historical review will be followed by a discussion of the recent development and future prospects of SWL technologies that may ultimately help to improve the clinical performance and safety of contemporary shock wave lithotripters. [Work supported by NIH through 5R37DK052985-18.]

WEDNESDAY AFTERNOON, 29 OCTOBER 2014 — INDIANA C/D, 2:00 P.M. TO 3:05 P.M.

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Uwe J. Hansen, Chair
Chemistry & Physics, Indiana State University, 64 Heritage Dr, Terre Haute, IN 47803-2374

Chair's Introduction—2:00

Invited Paper

2:05

3pED1. Educating mechanical engineers in the art of noise control. Colin Hansen (Mech. Eng., Univ. of Adelaide, 33 Parsons St., Marion, SA 5043, Australia, chansen@bigpond.net.au)

Acoustics and noise control is one of the disciplines where the material that students learn during a well-structured undergraduate course, can be immediately applied to many problems that they may encounter during their employment. However, in order to find optimal solutions to noise control problems, it is vitally important that students have a good fundamental understanding of the physical principles underlying the subject as well as a good understanding of how these principles may be applied in practice. Ideally, they should have access to affordable software and be confident in their ability to interpret and apply the results of any computer-based modelling that they may undertake. Students must fully understand any ethical issues that may arise, such as their obligation to ensure their actions do not contribute to any negative impact on the health and welfare of any communities. How do we ensure that our mechanical engineering graduates develop the understanding and knowledge required to tackle noise control problems that they may encounter after graduation? This presentation attempts to answer this question by discussing the process of educating undergraduate and postgraduate mechanical engineering students at the University of Adelaide, including details of lab classes, example problems, text books and software developed for the dual purpose of educating students and being useful in assisting graduates solve practical noise control problems.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Paul E. Barbone, Chair
Mechanical Engineering, Boston University, 110 Cummington St, Boston, MA 02215

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Online education: From classrooms to outreach, the internet is changing the way we teach and learn. Michael B. Wilson (Phys., North Carolina State Univ., 1649 Highlandon Ct, State College, PA 16801, wilsomb@gmail.com)

The internet is changing the face of education in the world today. More people have access to more information than ever before, and new programs are organizing and providing educational content for free to millions of internet users worldwide. This content ranges from interesting facts and demonstrations that introduce a topic to entire university courses. Some of these programs look familiar and draw from the media and education of the past, building off the groundwork laid by television programs like Watch Mr. Wizard, Bill Nye the Science Guy, and Reading Rainbow, with others more reminiscent of traditional classroom lectures. Some programs, on the other hand, are truly a product of modern internet culture and fan communities. 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 the state of acoustics in online education.

1:35

3pID2. Advanced methods of signal processing in acoustics. R. Lee Culver (School of Architecture, Rensselaer Polytechnic Inst., State College, Pennsylvania) and Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu)

Signal processing is applied in virtually all areas of modern acoustics to extract, classify, and/or quantify relevant information from acoustic measurements. Methods range from classical approaches based on Fourier and time-frequency analysis, to array signal processing, feature extraction, computational auditory scene analysis, and Bayesian inference, which incorporates physical models of the acoustic system under investigation together with advanced sampling techniques. This talk highlights new approaches to signal processing recently applied in a broad variety of acoustical problems.

2:05

3pID3. Hot topics in fish acoustics (active). Timothy K. Stanton (Dept. Appl. Ocean. Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, tstanton@whoi.edu)

It is important to quantify the spatial distribution of fish in their natural environment (ocean, lake, and river) and how the distribution evolves in time for a variety of applications including (1) management of fish stocks to maintain a sustainable source of food and (2) to improve our understanding of the ecosystem (such as how climate change impacts fish) through quantifying predator–prey relationships and other behavior. Active fish acoustics provides an attractive complement to nets given the great distances sound travels in the water and its ability to rapidly survey a large region at high resolution. This method involves studying distributions of fish in the water through analyzing their echoes through various means. While this field has enjoyed development for decades, there remain a number of "hot topics" receiving attention from researchers today. These include: (1) broadband acoustics as an emerging tool for advanced classification of, and discrimination between, species, (2) multi-beam imaging systems used to classify fish schools by size and shape, (3) long-range (km to 10's km) detection of fish, and (4) using transmission loss to classify fish on one-way propagation paths. Recent advances in these and other topics will be presented.

Session 3pNS

Noise: Sonic Boom and Numerical Methods

Jonathan Rathsam, Cochair
NASA Langley Research Center, MS 463, Hampton, VA 23681

Alexandra Loubeau, Cochair
NASA Langley Research Center, MS 463, Hampton, VA 23681

Contributed Papers

1:00

3pNS1. Source parameters for the numerical simulation of lightning as a nonlinear acoustic source. Andrew Marshall, Neal Evans, Chris Hackert, and Karl Oelschlaeger (Southwest Res. Inst., 6220 Culebra Rd., San Antonio, TX 78238-5166, andrew.marshall@swri.org)

Researchers have proposed using acoustic data to obtain additional insight into aspects of lightning physics. However, it is unclear how much information is retained in the nonlinear acoustic waveform as it propagates and evolves away from the lightning channel. Prior research in tortuous lightning has used simple N-waves as the initial acoustic emission. It is not clear if more complex properties of the lightning channel physics are also transmitted in the far-field acoustic signal, or if simple N-waves are a sufficient source term to predict far-field propagation. To investigate this, the authors have conducted a numerical study of acoustic emissions from a linear lightning channel. Using a hybrid strong-shock/weak-shock code, the authors compare the propagation of a simple N-wave and emissions from a source derived from simulated strong shock waves from the lightning channel. The implications of these results on the measurement of sound from nearby lightning sources will be discussed.

1:15

3pNS2. Nearfield acoustic measurements of triggered lightning using a one-dimensional microphone array. Maher A. Dayeh and Neal Evans (Southwest Res. Inst., Div 18, B77, 6220 Culebra Rd., San Antonio, TX 78238, neal.evans@swri.org)

For the first time, acoustic signatures from rocket-triggered lightning are measured by a 15 m long, one-dimensional microphone array consisting of 16 receivers, situated 79 m from the lightning channel. Measurements were taken at the International Center for Lightning Research and Testing (ICLRT) in Camp Blanding, FL, during the summer of 2014. We describe the experimental setup and report on the first observations obtained to date. We also discuss the implications of these novel measurements on the thunder initiation process and its energy budget during lightning discharges. Challenges of obtaining measurements in these harsh ambient conditions and their countermeasures will also be discussed.

1:30

3pNS3. The significance of edge diffraction in sonic boom propagation within urban environments. Jerry W. Rouse (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 North Dryden St., MS 463, Hampton, VA 23681, jerry.w.rouse@nasa.gov)

Advances in aircraft design, computational fluid dynamics, and sonic boom propagation modeling suggest that commercial supersonic aircraft can be designed to produce quiet sonic booms. Driven by these advances the decades-long government ban on overland supersonic commercial air transportation may be lifted. The ban would be replaced with a noise-based certification standard, the development of which requires knowledge of community response to quiet sonic booms. For inner city environments the estimation of community exposure to sonic booms is challenging due to the complex topography created by buildings, the large spatial extent and the required frequency range. Such analyses are currently intractable for traditional wave-based numerical methods such as the Boundary Element Method. Numerical methods based upon geometrical acoustics show promise, however edge diffraction is not inherent in geometrical acoustics and may be significant. This presentation shall discuss an initial investigation into the relative importance of edge diffraction in inner city sound fields caused by sonic booms. Results will provide insight on the degree to which edge diffraction effects are necessary for accurate predictions of inner city community exposure.

1:45

3pNS4. Sonic boom noise exposure inside homes. Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., MS 463, Hampton, VA 23681, j.klos@nasa.gov)

Commercial supersonic overland flight is presently banned both nationally and internationally due to the sonic boom noise that is produced in overflown communities. However, within the next decade, NASA and industry may develop and demonstrate advanced supersonic aircraft that significantly mitigate the noise perceived at ground level. To allow commercial operation of such vehicles, bans on commercial supersonic flight must be replaced with a noise-based certification standard. In the development of this standard, variability in the dose-response model needs to be identified. Some of this variability is due to differing sound transmission characteristics of homes both within the same community and among different communities. A tool to predict the outdoor-to-indoor low-frequency noise transmission into homes has been developed at Virginia Polytechnic Institute and State University, which was used in the present study to assess the indoor exposure in two communities representative of the northern and southern United States climate zones. Sensitivity of the indoor noise level to house geometry and material properties will be discussed. Future plans to model the noise exposure variation among communities within the United States will also be discussed.

2:00

3pNS5. Evaluation of the effect of aircraft size on indoor annoyance caused by sonic booms. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Sonic booms from recently proposed supersonic aircraft designs developed with advanced tools are predicted to be quieter than those from previous designs. The possibility of developing a low-boom flight demonstration vehicle for conducting community response studies has attracted international interest. These studies would provide data to guide development of a preliminary noise certification standard for commercial supersonic aircraft. An affordable approach to conducting these studies suggests the use of a

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sub-scale experimental aircraft. Due to the smaller size and weight of the sub-scale vehicle, the resulting sonic boom is expected to contain spectral characteristics that differ from that of a full-scale vehicle. To determine the relevance of using a sub-scale aircraft for community annoyance studies, a laboratory study was conducted to verify that these spectral differences do not significantly affect human response. Indoor annoyance was evaluated for a variety of sonic booms predicted for several different sizes of vehicles. Previously reported results compared indoor annoyance for the different sizes using the metric Perceived Level (PL) at the exterior of the structure. Updated results include analyses with other candidate noise metrics, nonlinear regression, and specific boom duration effects.

2:15

3pNS6. Effects of secondary rattle noises and vibration on indoor annoyance caused by sonic booms. Jonathan Rathsam (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jonathan.rathsam@nasa.gov)

For the past 40 years, commercial aircraft have been banned from overland supersonic flight due to the annoyance caused by sonic booms. However, advanced aircraft designs and sonic boom prediction tools suggest that significantly quieter sonic booms may be achievable. Additionally, aircraft noise regulators have indicated a willingness to consider replacing the ban with a noise-based certification standard. The outdoor noise metric used in the certification standard must be strongly correlated with indoor annoyance. However, predicting indoor annoyance is complicated by many factors including variations in outdoor-to-indoor sound transmission and secondary indoor rattle noises. Furthermore, direct contact with vibrating indoor surfaces may also affect annoyance. A laboratory study was recently conducted to investigate candidate noise metrics for the certification standard. Regression analyses were conducted for metrics based on the outdoor and transmitted indoor sonic boom waveforms both with and without rattle noise, and included measured floor vibration. Results indicate that effects of vibration are significant and independent of sound level. Also, the presence or absence of rattle sounds in a transmitted sonic boom signal generally changes the regression coefficients for annoyance models calculated from the outdoor sound field, but may not for models calculated from the indoor sound field.

2:30

3pNS7. Artificial viscosity in smoothed particle hydrodynamics simulation of sound interference. Xu Li, Tao Zhang, YongOu Zhang (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei Province 430074, China, lixu199123@gmail.com), Huajiang Ouyang (School of Eng., Univ. of Liverpool, Liverpool, United Kingdom), and GuoQing Liu (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei Province, China)

The artificial viscosity has been widely used in reducing unphysical oscillations in the Smoothed Particle Hydrodynamics (SPH) simulations. However, the effects of artificial viscosity on the SPH simulation of sound interference have not been discussed in existing literatures. This paper analyzes the effects and gives some suggestions on the choice of computational parameters of the artificial viscosity in the sound interference simulation. First, a standard SPH code for simulating sound interference in the time domain is built by solving the linearized acoustic wave equations. Second, the Monaghan type artificial viscosity is used to optimize the SPH simulation. Then the SPH codes with and without the artificial viscosity are both used to simulate the sound interference and the numerical solutions are compared with the theoretical results. Finally, different values of computational parameters of the artificial viscosity are used in the simulation in order to determine the appropriate values. It turns out that the numerical solutions of SPH simulation of sound interference agree well with the theoretical results. The artificial viscosity can improve the accuracy of the sound interference simulation. The appropriate values of computational parameters of the artificial viscosity are recommended in this paper.

2:45

3pNS8. Smoothed particle hydrodynamics simulation of sound reflection and transmission. YongOu Zhang (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan 430074, China, zhangyo1989@gmail.com), Tao Zhang (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei Province, China), Huajiang Ouyang (School of Eng., Univ. of Liverpool, Liverpool, United Kingdom), and TianYun Li (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, China)

Mesh-based methods are widely used in acoustic simulations nowadays. However, acoustic problems with complicated domain topologies and multiphase systems are difficult to be described with these methods. On the contrary, Smoothed Particle Hydrodynamics (SPH), as a Lagrangian method, does not have much trouble in solving these problems. The present paper aims to simulate the reflection and transmission of sound waves with the SPH method in time domain. Firstly, the linearized acoustic equations are represented in the SPH form by using the particle approximation. Then, one dimensional sound reflection and transmission are simulated with the SPH method and the solutions are compared with the theoretical results. Finally, the effects of smoothing length and neighboring particle numbers on the computation are discussed. The errors of sound pressure, particle velocity, and change of density show that the SPH method is feasible in simulating the reflection and transmission of sound waves. Meanwhile, the relationship between the characteristic impedance and the reflected waves obtained by the SPH simulation is consistent with the theoretical result.

3:00

3pNS9. A high-order Cartesian-grid finite-volume method for aeroacoustics simulations. Mehrdad H. Farahani (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., UCLA School of Medicine, 1000 Veteran Ave., Los Angeles, CA 90095, mh.farahani@gmail.com), John Mousel (Mech. and Industrial Eng., The Univ. of Iowa, Iowa City, IA), and Sarah Vigmostad (Biomedical Eng., The Univ. of Iowa, Iowa City, IA)

A moving-least-square based finite-volume method is developed to simulate acoustic wave propagation and scattering from complicated solid geometries. This hybrid method solves the linearized perturbed compressible equations as the governing equations of the acoustic field. The solid boundaries are embedded in a uniform Cartesian grid and represented using level set fields. Thus, the current approach avoids unstructured grid generation for the irregular geometries. The desired boundary conditions are imposed sharply on the immersed boundaries using a ghost fluid method. The scope of the implementation of the moving moving-least-square approach in the current solver is threefold: reconstruction of the field variables on cell faces for high-order flux construction, population of the ghost cells based on the desired boundary condition, and filtering the high wave number modes near the immersed boundaries. The computational stencils away from the boundaries are identical; hence, only one moving-least-square shape-function is computed and stored with its underlying grid pattern for all the interior cells. This feature significantly reduces the memory requirement of the acoustic solver compared to similar finite-volume method on irregular unstructured mesh. The acoustic solver is validated against several benchmark problems.

Session 3pUW

Underwater Acoustics: Shallow Water Reverberation I

Dajun Tang, Chair
Applied Physics Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Chair's Introduction—1:00

Invited Papers

1:05

3pUW1. Overview of reverberation measurements in Target and Reverberation Experiment 2013. Jie Yang, Dajun Tang, Brian T. Hefner, Kevin L. Williams (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jieyang@apl.washington.edu), and John R. Preston (Appl. Res. Lab., Penn State Univ., State College, PA)

The Target and REverberation EXperiment 2013 (TREX13) was carried out off the coast of Panama City, Florida, from 22 April to 16 May, 2013. Two fixed-source/fixed-receiver acoustic systems were used to measure reverberation over time under diverse environmental conditions, allowing study of reverberation level (RL) dependence on bottom composition, sea surface conditions, and water column properties. Beamformed RL data are categorized to facilitate studies emphasizing (1) bottom reverberation; (2) sea surface impact; (3) biological impact; and (4) target echo. This presentation is an overview of RL over the entire experiment, summarizing major observations and providing a road map and suitable data sets for follow-up efforts on model/data comparisons. Emphasis will be placed on the dependence of RL on local geoacoustic properties and sea surface conditions. [Work supported by ONR.]

1:25

3pUW2. Non-stationary reverberation observations from the shallow water TREX13 reverberation experiments using the FORA triplet array. John R. Preston (ARL, Pennsylvania State Univ., P. O. Box 30, MS3510, State College, PA 16804, jrp7@arl.psu.edu), Douglas A. Abraham (CausaSci LLC, Ellicott City, MD), and Jie Yang (APL, Univ. of Washington, Seattle, WA)

A large experimental effort called TREX13 was conducted in April-May 2013 off Panama City, Florida. As part of this effort, reverberation and clutter measurements were taken in a fixed-fixed configuration in very shallow water (~20 m) over a 22 day period. Results are presented characterizing reverberation, clutter and noise in the 1800-5000 Hz band. The received data are taken from the triplet subaperture of the Five Octave Research Array (FORA). The array was fixed 2 m off the sea floor and data were passed to a nearby moored ship (the R/V Sharp). An ITC 2015 source transducer was fixed 1.1 m off the seafloor nearby. Pulses comprised of gated CWs and LFMs were used in this study. Matched filtered polar plots of the reverberation and clutter are presented using the FORA triplet beamformer. There are clear indications of biologic scattering. Some of the nearby shipwrecks are clearly visible in the clutter, as are reflections from a DRDC air-filled hose. The noise data show a surprising amount of time-dependent anisotropy. Some statistical characterization of these various components of the reverberation are presented using K-distribution based algorithms to note differences in the estimated shape parameter. Help from the Applied Physics Laboratory at the University of Washington was crucial to this effort. [Work supported by ONR code 322OA.]

Contributed Paper

1:45

3pUW3. Propagation measurement using source tow and moored vertical line arrays during TREX13. William S. Hodgkiss, David Ensberg (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA), and Dajun Tang (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

The objective of TREX13 (Target and Reverberation EXperiment 2013) is to investigate shallow water reverberation by concurrently measuring propagation, local backscatter, and reverberation, as well as sufficient environmental parameters needed to achieve unambiguous model/data comparison. During TREX13 the Marine Physical Laboratory (MPL) conducted propagation and forward scatter measurements. The MPL effort during TREX13 included deploying three, 32-element (0.2 m element spacing), vertical line arrays along the Main Reverberation Track at a bearing of ~128° and ranges ~2.4 km, ~4.2 km, and ~0.5 km from the R/V Sharp, where reverberation measurements were being made. In addition, MPL carried out repeated source tows in the band of 2–9 kHz along the Main Reverberation Track, using tonal and LFM waveforms. The experimental procedure is described and the resulting source-tow data is examined in the context of Transmission Loss and its implications for reverberation.

Invited Papers

2:00

3pUW4. Comparison of signal coherence for continuous active and pulsed active sonar measurements in littoral waters. Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, phines50@gmail.com), Stefan M. Murphy (Defence R&D Canada, Dartmouth, NS, Canada), and Keaton T. Hicks (Dept. of Mech. Eng., Dalhousie Univ., Halifax, NS, Canada)

Military sonars must detect, localize, classify, and track submarine threats from distances safely outside their circle of attack. However, conventional pulsed active sonars (PAS) have duty cycles on the order of 1% which means that 99% of the time, the track is out of date. In contrast, continuous active sonars (CAS) have a 100% duty cycle, which enables continuous updates to the track. This should significantly improve tracking performance. However, one would typically want to maintain the same bandwidth for a CAS system as for the PAS system it might replace. This will provide a significant increase in the time-bandwidth product, but may not produce the increase in gain anticipated if there are coherence limitations associated with the acoustic channel. To examine the impact of the acoustic channel on the gain for the two pulse types, an experiment was conducted as part of the Target and Reverberation Experiment (TREX) in May 2013 using a moored active sonar and three passive acoustic targets, moored at ranges from 2 to 6 km away from the sonar. In this paper, preliminary results from the experiment will be presented. [Work supported by the U.S. Office of Naval Research.]

2:20

3pUW5. Reverberation and biological clutter in continental shelf waveguides. Ankita D. Jain, Anamaria Ignisca (Mech. Eng., Massachusetts Inst. of Technol., Rm. 5-229, 77 Massachusetts Ave., Cambridge, MA 02139, ankitadj@mit.edu), Mark Andrews, Zheng Gong (Elec. & Comput. Eng., Northeastern Univ., Boston, MA), Dong Hoon Yi (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Purnima Ratilal (Elec. & Comput. Eng., Northeastern Univ., Boston, MA), and Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

Seafloor reverberation in continental shelf waveguides is the primary limiting factor in active sensing of biological clutter in the ocean for noise unlimited scenarios. The detection range of clutter is determined by the ratio of the intensity of scattered returns from clutter versus the seafloor in a resolution cell of an active sensing system. We have developed a Rayleigh-Born volume scattering model for seafloor scattering in an ocean waveguide. The model has been tested with data collected from a number of Ocean Acoustic Waveguide Remote Sensing (OAWRS) experiments in distinct US Northeast coast continental shelf environments, and has shown to provide accurate estimates of seafloor reverberation over wide areas for various source frequencies. We estimate scattered returns from fish clutter by combining ocean-acoustic waveguide propagation modeling that has been calibrated in a variety of continental shelf environments for OAWRS applications with a model for fish target strength. Our modeling of seafloor reverberation and scattered returns from fish clutter is able to explain and elucidate OAWRS measurements along the US Northeast coast.

Contributed Papers

2:40

3pUW6. Transmission loss and reverberation variability during TREX13. Sean Pecknold (DRDC Atlantic Res. Ctr., PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca), Diana McCammon (McCammon Acoust. Consulting, Waterville, NS, Canada), and Dajun Tang (Ocean Acoust., Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The ONR-funded Target and Reverberation Experiment 2013 (TREX13) took place in the Northeastern Gulf of Mexico near Panama City, Florida, during April and May of 2013. During this trial, which took place in a shallow water (20 m deep) environment, several sets of one-way and two-way acoustic transmission loss and reverberation data were acquired. Closed form expressions are derived to trace the uncertainty in the inputs to a Gaussian beam propagation model through the model to obtain an estimate of the uncertainty in the output, both for transmission loss and for reverberation. The measured variability of the TREX environment is used to compute an estimate of the expected transmission loss and reverberation variability. These estimates are then compared to the measured acoustic data from the trial.

2:55

3pUW7. Transmission loss and direction of arrival observations from a source in shallow water. David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng. Dept., Univ. of Washington, Seattle, WA)

Signals generated by the source used in the reverberation studies of the Targets and Reverberation Experiment (TREX) were recorded by a receiving array located 4.7 km downrange. The bathymetry over this range is relatively flat, with water depth 20 m. The receiving system consists of a 7-channel vertical line array, a 4-channel horizontal line array, oriented perpendicular to the propagation direction, and a 4-channel vector sensor (3-component vector and one pressure), with all channels recorded coherently recorded Transmissions were made once every 30 seconds and over a two hour recording period, changes in the frequency content, amplitude and direction were observed. As both the source and receiving array are at a fixed position in the water column, these observations are assumed to be due to changes in the environment. Interpretation of the data is given in terms of the evolving sea-surface conditions, the presence of nearby scatterers such as fish, and reflection/refraction due to the sloping shoreline.

3:10

3pUW8. Effect of a roughened sea surface on shallow water propagation with emphasis on reactive intensity obtained with a vector sensor. David R. Dall'Osto (Appl. Phys. Lab., Univ. Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Appl. Phys. Lab. and Mechanical Eng. Dept., Univ. of Washington, Seattle, WA)

In this study, sea-surface conditions during the Targets and Reverberation Experiment (TREX) are analyzed. The sea-surface directional spectrum was experimentally measured up to 0.6 Hz with two wave buoys separated by 5 km. The analysis presented here focuses on propagation relating to three canonical sea-surfaces observed during the experiment: calm conditions, and rough conditions with waves either perpendicular or parallel to the primary propagation direction. Acoustic data collected during calm and rough conditions show a significant difference in the amount of out-of-plane scattering. Interference due to this out-of-plane scattering is observed in the component of reactive intensity perpendicular to the propagation direction. These observations are compared with those generated using a model of the sea-surface scattering based on a combination of buoy-measured and modeled directional spectrum. Simulated sea-surfaces are also constructed for this numerical study. A model for wind waves is used to obtain surface wavenumbers greater than those measured by the wave buoys (~1.5 rad/m). Importantly, the spectral peak and it direction are well measured by the buoys and no assumptions on fetch are required, resulting in a more realistic wave spectrum and description of sea-surface conditions for acoustic modeling.

Plenary Session, Annual Meeting, and Awards Ceremony

Judy R. Dubno, President
Acoustical Society of America

Annual Meeting of the Acoustical Society of America

Presentation of Certificates to New Fellows

Mingsian Bai – for contributions to nearfield acoustic holography

David S. Burnett – for contributions to computational acoustics

James E. Phillips – for contributions to vibration and noise control and for service to the Society

Bonnie Schnitta – for the invention and application of noise mitigation systems

David R. Schwind – for contributions to the acoustical design of theaters, concert halls, and film studios

Neil T. Shade – for contributions to education and to the integration of electroacoustics in architectural acoustics

Joseph A. Turner – for contributions to theoretical and experimental ultrasonics

Announcements and Presentation of Awards

Presentation to Leo L. Beranek on the occasion of his 100th Birthday

Rossing Prize in Acoustics Education to Colin H. Hansen

Pioneers of Underwater Acoustics Medal to Michael B. Porter

Silver Medal in Speech Communication to Sheila E. Blumstein

Wallace Clement Sabine Medal to Ning Xiang

WEDNESDAY EVENING, 30 OCTOBER 2014 7:30 P.M. TO 9:00 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. On Tuesday the meetings will begin at 8:00 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m. On Wednesday evening, the Technical Committee on Biomedical Acoustics will meet starting at 7:30 p.m. On Thursday evening, the meetings will begin at 7:30 p.m.

Biomedical Acoustics Indiana A/B

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 meetings participants are cordially invited to attend these meetings and to participate actively in the discussion.

ACOUSTICAL SOCIETY OF AMERICA
PIONEERS OF UNDERWATER ACOUSTICS MEDAL

Michael B. Porter

2014

The Pioneers of Underwater Acoustics Medal is presented to an individual irrespective of nationality, age, or society affiliation, who has made an outstanding contribution to the science of underwater acoustics, as evidenced by publication of research in professional journals or by other accomplishments in the field. The award was named in honor of five pioneers in the field: H. J. W. Fay, R. A. Fessenden. H. C. Hayes, G. W. Pierce, and P. Langevin.

PREVIOUS RECIPIENTS

Harvey C. Hayes	1959	Robert J. Urick	1988
Albert B. Wood	1961	Ivan Tolstoy	1990
J. Warren Horton	1963	Homer P. Bucker	1993
Frederick V. Hunt	1965	William A. Kuperman	1995
Harold L. Saxton	1970	Darrell R. Jackson	2000
Carl Eckart	1973	Frederick D. Tappert	2002
Claude W. Horton, Sr.	1980	Henrik Schmidt	2005
Arthur O. Williams	1982	William M. Carey	2007
Fred N. Spiess	1985	George V. Frisk	2010

CITATION FOR MICHAEL B. PORTER

. . . for contributions to underwater acoustic modeling

INDIANAPOLIS, INDIANA • 29 OCTOBER 2014

Michael B. Porter describes his college days as "ending up" at Caltech, living in an eleven-student communal environment, sleeping on a floor-mattress, working in various odd-jobs and mastering the culinary skill of baking beans, all in anticipation of his immediate future that would be dominated by his student loan. Aside from what to him were his major undergraduate accomplishments, meeting his significant other, Laurel Henderson, and developing crowd pleasing culinary talents, he also apparently learned some math and physics. At Northwestern, he received his Ph.D. in Applied Math working for Ed Reiss in which, among other things, he developed numerical algorithms that were to become standard methods in the Underwater Acoustics (UW) community. Probably, of the four most often used models in all of the UW community in the last quarter century or so, Michael B. Porter is the originator of two of them: The KRAKEN normal mode models and BELLHOP, a ray-Gaussian beam model.

However, these major contributions were only a few along the diverse research trail that Michael pioneered. His research venues were also as diverse in that his 35 years of research were conducted while in research, professor, and management positions at more or less every type of research organization—government, academic, and private sector. His research community impact is pervasive in that he is a coauthor of *Computational Ocean Acoustics*, recently revised in a second edition, and he has also created and maintains the Ocean Acoustics Library (OALIB), a site where anyone can download his MATLAB versions of all the major underwater acoustic propagation models. These latter activities alone probably make Mike Porter a "household name" for the whole international community in UW—but these are only part of the story.

Mike's first pioneering acoustic contribution made not too long after he was born in Quebec City in 1958 was when he developed an innovative glue-based repair procedure for woofer-speakers that produced flying bits of speaker cone when directly plugged into a wall outlet. While this experience probably motivated him to explore numerical methods, he did spend some time later working on transducers with George Benthien at the Naval Ocean Systems Center (NOSC).

His groundbreaking Ph.D. thesis and seminal paper in 1984 in the *Journal of the Acoustical Society of America* (JASA) on an unconditionally stable approach to normal mode computation laid the foundation for KRACKEN/KRACKEN C and the SACLANTCEN SNAP normal mode (NM) models, probably the two most used NM models in the world. After his Ph.D. he was fortunate enough to collaborate with Homer Bucker at NOSC on Gaussian beams from which BELLHOP would be an outgrowth. So, almost immediately after his Ph.D. he was a leader in the UW modeling community.

I first met Mike when he came to the Naval Research Laboratory (NRL) in 1985 to work in Orest Diachok's Branch on Arctic (and other) acoustics. There he further organized his models to community useable tools while also significantly contributing to the new area of Matched Field Processing (MFP). We also established a close working relationship in that area and in particular worked on a rapid method to do three-dimensional modal propagation, which he later enhanced to include more oceanographic as well as global propagation phenomena. We have remained close friends and colleagues since those NRL days, including coauthoring *Computational Ocean Acoustics* with Finn Jensen and Henrik Schmidt.

In 1987 Mike joined Finn Jensen's modeling group at SACLANTCEN, and it was there that he worked with the rest of the coauthors (all from or at SACLANTCEN) on *Computational Ocean Acoustics*. There he also developed ongoing research partnerships with U. S. oceanographer Steve Piacsek as well as other European scientists. Much of his SACLANTCEN research concerned range-dependent modeling, including a seminal contribution to energy conservation of one-way equations, coupled mode modeling, and chaotic effects in multipath environments.

He returned to the U. S. in 1991 to a faculty position at the New Jersey Institute of Technology's (NJIT) math department with David Stickler, Daljit Ahluwahlia, and Greg Kriegsmann, and was rather quickly elevated to being one of its youngest full professors.

There he worked in the area of MFP, extending it to some complicated broadband scenarios (with Zoi-Heleni Michalopoulou) as well further optimizing his models. While at NJIT, he also did a sabbatical at the University of Algarve with his former SACLANTCEN colleague Sergio Jesus and with the Portuguese and French hydrographers Yann Stephan and Emanuel Coelho to study the acoustic effects of internal tides.

In 1999 he accepted a position at Science Applications International Corporation (SAIC) as Assistant Vice President/Chief Scientist in its Ocean Science Division headed by Peter Mikhalevsky. It was there that he began close collaborations with Paul Hursky, Ahmad Abawi, Martin Siderius, and Keyko McDonald (from SPAWAR). At SAIC he also completed his transition to heavy-duty experimental activity, which probably originated by him being misled at SACLANTCEN to think that at-sea experiments were associated with Michelin rated dining. His subsequent growth as an at-sea scientist is evidenced by his role as chief scientist on a series of multi-institutional acoustic communications (Acomms) sea trials. Mike was uniquely qualified for this Acomms role in that the only practical model to describe the Acomms channel was BELLHOP. So, as the experiment chief scientist he was also the expert on the theoretical aspects of the project. He had progressed to a level that made him one of a very few scientists in our community capable of leading the theory, simulation, and experimental aspects of a large UW project. This was all happening while he was also working in MFP and inverse methods at SAIC.

Ever restless and seeking new experiences, he founded a new company in 2004, Heat Light and Sound, Inc. (HLS), taking with him Abawi, Hursky, and Siderius. At HLS he has continued his research, lately being involved in ocean soundscapes, marine mammal acoustics, and other environmentally-related areas as well as continuing on in his established fields of research. During this latter period he was also a coauthor of the seminal paper in JASA (2006) on the passive fathometer with Martin Siderius and Chris Harrison.

Most important to me is that Mike has been my very good friend over these many years, and it has been a pleasure to watch him share his friendship and his knowledge with a very broad segment of the UW community. He is an author of acoustic models and a book that are central to the acoustic community and has established the Ocean Acoustics Library, the latter probably being the most important instrument in disseminating models to students as well as seasoned researchers. Recognized early in his career with the A. B. Wood Medal, Michael B. Porter's career trajectory in Underwater Acoustics has truly been a pioneering adventure. The ASA Pioneers of Underwater Acoustics Medal is a fitting recognition of his many achievements.

WILLIAM A. KUPERMAN

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Speech Communication

Sheila E. Blumstein

2014

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

Franklin S. Cooper	1975
Gunnar Fant	1980
Kenneth N. Stevens	1983
Dennis H. Klatt	1987
Arthur S. House	1991
Peter Ladefoged	1994
Patricia K. Kuhl	1997
Katherine S. Harris	2005
Ingo R. Titze	2007
Winifred Strange	2008
David B. Pisoni	2010

CITATION FOR SHEILA E. BLUMSTEIN

. . . for contributions to understanding how acoustic signals are transformed into linguistic representations

INDIANAPOLIS, INDIANA • 29 OCTOBER 2014

Sheila Blumstein was born in New York City, obtained a B.A. in Linguistics from the University of Rochester, and a Ph.D. in Linguistics from Harvard University, under the guidance of the legendary Roman Jakobson. Sheila's dissertation, *A Phonological Investigation of Aphasic Speech*, published as a book by Mouton in 1973, already clearly indicated the focus of her research: the representation of speech and language in the brain. Today, as the Albert D. Mead Professor of Cognitive and Linguistic Sciences at Brown University, Sheila pursues this research agenda as vigorously as when she started there on the faculty in 1970.

Sheila Blumstein has contributed immeasurably to our knowledge of the acoustics and perception of speech. Specifically, her research addresses how the continuous acoustic signal is transformed by perceptual and neural mechanisms into linguistically relevant representations. Among her many significant contributions to the field of Speech Communication, the following two have had a profound impact on our field. First, through detailed analysis of speech sounds, Sheila showed that the mapping between acoustic properties and perceived phonetic categories is richer, and more consistent and invariant, than previously thought, a finding which necessitated a new conception of the relation between the production and perception of speech. Second, Sheila's finding that subtle yet systematic acoustic differences can affect activation of word candidates in the mental lexicon indicated that acoustic information not directly relevant for phoneme identification is not discarded but is retained and plays a critical role in word comprehension, providing a crucial piece of evidence in the ongoing debate about the structure of the mental lexicon.

At the time that Sheila started investigating the speech signal in the 1970s, the prevalent scientific opinion was that there was no simple mapping between acoustic signal and perceived phonemes because the speech signal was too variable. Acoustic properties were strongly affected by contextual factors such as variation in speaker, speaking rate, and phonetic environment. Careful consideration of Gunnar Fant's acoustic theory of speech production led Sheila to the hypothesis that invariant acoustic properties could be found in the speech signal. In contrast to previous research that was dependent on the speech spectrograph, Sheila focused more on global acoustic properties such as the overall shape of the spectrum at the release of the stop consonant. Through careful and detailed acoustic analysis and subsequent perceptual verification, Sheila uncovered stable invariant acoustic properties that consistently signaled important linguistic features such as place and manner of articulation. Sheila supported these claims by investigating a variety of speech sound classes (including stop consonants, fricatives, and approximants) in a variety of languages because she fully appreciated that conclusions drawn on the basis of one language can be misleading and universal generalizations can only be made after crosslinguistic comparisons. Sheila's work on acoustic features resulted in a series of seminal publications (1978-1987) in the *Journal of the Acoustical Society of America*, co-authored with Kenneth Stevens and others.

By the late 1980s, research on speech perception had moved beyond the identification of individual consonants and vowels to the comprehension of words and to the new field of "auditory word recognition." While there was a general consensus that word recognition involves a process whereby information extracted from the speech signal is matched with a stored representation in the mental lexicon, it was not clear whether all available acoustic information in the signal played a role in this matching process. In her seminal paper "The effect of subphonetic differences on lexical access" (*Cognition*, 1994), Sheila and her students showed that subtle acoustic variations which do not affect the categorization of a phoneme nevertheless do affect word recognition. This was a very elegant demonstration that subtle subphonemic acoustic information is not discarded before the lexicon is accessed but instead plays a role in the comprehension of words. This was a very important finding and necessitated reconsideration of the then dominant view that lexical access proceeds on the basis of categorical phonemes rather than more fine-grained continuous acoustic information.

Sheila is co-founder of Brown University's Barus Speech Lab where she has taught, supervised, and mentored hundreds of undergraduates, graduates, and postdocs. This lab is one of the world's leading research centers for the study of speech at all levels: acoustics, psycholinguistics, and neurolinguistics. In addition to her speech research, Sheila is equally known for her research on aphasia, focusing again on speech production and perception. Just as Sheila was able to make use of technological advances to view the speech signal from a different perspective, she also capitalized on new brain imaging techniques to augment her understanding of the brain that was based on behavioral data collected from aphasic patients. Sheila's most recent acoustic research also uses fMRI to investigate cortical regions involved in the perception of phonetic category invariance as well as neural systems underlying lexical competition.

A quick glance at Sheila's resume shows that she has garnered just about every honor possible. She has been a Guggenheim Fellow, and a recipient of the Claude Pepper (Javits Neuroscience) Investigator Award. She is a Fellow of the Acoustical Society of America, the American Association for the Advancement of Science, the American Academy of Arts and Sciences, the Linguistic Society of America, and the American Philosophical Society. In addition, Sheila has served Brown University in many capacities, including Dean of the College, Interim Provost, and Interim President. In all of the positions she has held, Sheila has earned the admiration and respect of all constituencies. Her warm, supportive, patient style renders an incisive critique into a constructive suggestion, reflecting her enviable supervisory and administrative skills.

It is simply not possible to undertake work in acoustic phonetics, phonology, neuroimaging, or aphasia without referring to Sheila's work. Sheila's research has been continuously funded through federal research grants since the 1970s. Her research is not only influential and pivotal, it is also incredibly inspiring. Her students have secured prestigious positions and continue to conduct innovative research. The field would not be what it is today without Sheila's many seminal contributions spanning five decades.

ALLARD JONGMAN
JOAN SERENO
SHARI BAUM
ADITI LAHIRI

WALLACE CLEMENT SABINE AWARD
OF THE
ACOUSTICAL SOCIETY OF AMERICA

Ning Xiang

2014

The Wallace Clement Sabine Award is presented to an individual of any nationality who has furthered the knowledge of architectural acoustics, as evidenced by contributions to professional journals and periodicals or by other accomplishments in the field of architectural acoustics.

PREVIOUS RECIPIENTS

Vern O. Knudsen	1957
Floyd R. Watson	1959
Leo L. Beranek	1961
Erwin Meyer	1964
Hale J. Sabine	1968
Lothar W. Cremer	1974
Cyril M. Harris	1979
Thomas D. Northwood	1982
Richard V. Waterhouse	1990
A. Harold Marshall	1995
Russell Johnson	1997
Alfred C. C. Warnock	2002
William J. Cavanaugh	2006
John S. Bradley	2008
J. Christopher Jaffe	2011

SILVER MEDAL IN ARCHITECTURAL ACOUSTICS

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 RECIPIENT

Theodore J. Schultz 1976

CITATION FOR NING XIANG

. . . for contributions to measurements and analysis techniques, and numerical simulation of sound fields in coupled rooms

INDIANAPOLIS, INDIANA • 29 OCTOBER 2014

Ning Xiang, 16th recipient of the Society's Wallace Clement Sabine Medal, is well known to members of the Society and the worldwide acoustics community for his work in binaural scale-model measurement, theory and practice of maximum-length sequences, and Bayesian signal processing. A consummate theoretician and experimentalist, his work reflects the growing importance of computational modeling and model-based signal processing across the broader field of acoustics, but is unique for making significant general contributions while maintaining a strong and specific focus on architectural acoustics.

Ning formally began his career in acoustics in 1984, arriving as a young student from China at the office of his doctoral supervisor Jens Blauert. Though more-or-less inexperienced in the field and hardly able to communicate in German, his mentors and colleagues of that time well remember his fierce determination and commitment. This earnest enthusiasm for the work would serve him well over his professional career, becoming one of the key attributes he sought to instill in the many graduate students he would come to supervise.

Earning a Masters degree in 1986 (Diplom-Ingenieur) from Ruhr-University Bochum, Ning went on to earn a Ph.D. in 1990 for his development of a binaural acoustical modeling system. This work, which involved design and fabrication of novel scale-model transducers and a miniature (1/10 scale) binaural artificial head, set early the high standard his future experimental work would demonstrate. At the same time, his doctoral work firmly established him as a theorist and signal processor for his research and development of measurement algorithms and software based on maximum-length sequences. This included a new and effective factorization method required for application of Fast Hadamard Transforms and development of fast test methods for long maximum-length sequences through identification of the similarity with Morse-Thue sequences [*Signal Processing* (1992)]. It was through these important findings in maximum-length sequences that Ning began a long and fruitful collaborative relationship with Manfred Schroeder [*Journal of the Acoustical Society of America* (JASA) (2003)].

After the completion of his doctoral degree, Ning joined the technical staff of Head acoustics in Herzogenrath, Germany as a research scientist/engineer. Here too he continued to bring together theory and practice and experimental work with signal processing, forming an on-going and fruitful professional relationship with founder Klaus Genuit that led to a number of important papers [JASA (1995), *ACUSTICA - acta acustica* (1996)].

This was followed by an appointment in 1997 as a research scientist at the Fraunhofer Institute for Building Physics in Stuttgart, Germany. Here the application of binaural measurement technology to performance spaces remained his focus. While this was to be Ning's last appointment in Germany, his many professional relationships remain strong and he is well remembered by his colleagues for his ability to appealingly distill in his lectures and talks the rigor of his analytical thinking into well-organized, clearly articulated concepts without sacrificing substance or detail.

In 1998 Ning accepted a position as a Research Scientist and Research Associate Professor with the National Center for Physical Acoustics and the Department of Electrical Engineering of the University of Mississippi. His work on acoustic/seismic coupling for buried mine detection, conducted in collaboration with James Sabatier and Paul Goggans, was a departure in domain from his prior work in room and building acoustics. But, characteristically, it became for Ning an opportunity for fertile cross-pollination between subdisciplines. Advances he had made in measurement by maximum-length sequence transitioned into acoustic/seismic measurement while advances in Bayesian signal processing for mine detection provided him with an important new approach for parameters estimation from single- and multiple-slope Schroeder decay curves of noisy impulse responses [JASA (2001, 2003)].

In 2003, Ning was appointed Associate Professor at the Rensselaer Polytechnic Institute (RPI). Returning his full attention to the field of architectural acoustics, Ning expanded his on-going work in maximum-length sequences and Bayesian estimation. His work in parameter and model estimation for systems of acoustically coupled rooms led him naturally into development of new computational diffusion-equation models for simulation of acoustically coupled rooms and detailed scale-model measurements to validate these models. From this work—much of it carried out with his Masters and Ph.D. students and conducted with a worldwide group of collaborators—has grown prodigiously and now encompasses modeling, measurement, and simulation of scattering, material impedance, and mode distribution in addition to binaural measurement and multiple-slope decay curve analysis.

It is especially fitting that Ning should receive this award directly after J. Christopher Jaffe, as Ning has been instrumental in bringing to full fruition the work begun by Dr. Jaffe in founding the Graduate Program in Architectural Acoustics at RPI in 1999 and the research in coupled rooms that was the initial focus of that program [JASA (2005, 2006, 2008, 2009, 2011, 2013)]. The flourishing and growth of the RPI program directed by Ning since 2005 is, along with his other scholarly and professional accomplishments, one of his enduring contributions to the field of architectural acoustics.

This educational and mentoring role cannot be overemphasized; in the close community of architectural and room acoustics Ning's direct role in training a new generation of acousticians has been felt across academia, government, and industry in both the U.S. and abroad. Whether in consulting practice in Turkey, as a Fulbright fellow in Finland, or as a professor in the United States, a community of young acousticians is daily reaping benefits of having been educated to pursue the field of architectural acoustics with scientific and engineering rigor coupled with a bold willingness to investigate new ideas and an openness to the worldwide acoustics community. Doubtless Ning recalls his own enthusiasm as a young graduate student in Bochum and seeks to cultivate that in his own students.

For the reasons cited here and those which space does not allow us to mention but are well known to his colleagues, students, and the Society as a whole, we are pleased and privileged to present Dr. Ning Xiang with the Wallace Clement Sabine Medal.

JASON E. SUMMERS
JENS BLAUERT

Session 4aAAa

Architectural Acoustics, Speech Communication, and Noise: Room Acoustics Effects on Speech Comprehension and Recall I

Lily M. Wang, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

David H. Griesinger, Cochair
Research, David Griesinger Acoustics, 221 Mt Auburn St #504, Cambridge, MA 02138

Chair's Introduction—8:40

Invited Papers

8:45

4aAAa1. Speech recognition in adverse conditions. Ann Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

Speech recognition is highly sensitive to adverse conditions at all stages of the speech chain, i.e., the sequence of events that transmits a message from the mind/brain of a speaker through the acoustic medium to the mind/brain of a listener. Adverse conditions can originate from source degradations (e.g., disordered or foreign-accented speech), environmental disturbances (e.g., background sounds with or without energetic masking), and/or receiver (i.e., listener) limitations (e.g., impaired or incomplete language models, peripheral deficiencies, or tasks with high cognitive load). (For more on this classification system, see Mattys, Davis, Bradlow, & Scott, 2012, Language and Cognitive Processes, 27). This talk will present a series of studies focused on linguistic aspects of these various possible sources of adverse conditions for speech recognition. In particular, we will demonstrate separate and combined influences of the talker's language background (a possible source degradation), the presence of a background speech masker in either the same or a different language from that of the target speech (a possible environmental degradation), and the listener's experience with the language of the target and/or masking speech (a possible receiver limitation). Together, these studies demonstrate strong influences of language and linguistic experience on speech recognition in adverse conditions.

9:05

4aAAa2. Speech intelligibility and sentence recognition memory in noise. 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 in adverse conditions impacting various levels of speech processing negatively. These adverse conditions may originate in talker- (fast, reduced speech), signal- (noise or degraded target signal), and listener- (impeded access or decoding of the target speech signal) oriented limitations, and may have consequences for perceptual processes, representations, attention, and memory functions (see Mattys *et al.*, 2012 for a review). In this talk, I first discuss a set of experiments that explore the extent to which listener-oriented clear speech and speech produced in response to noise (noise-adapted speech) by children, young adults and older adults contribute to enhanced word recognition in challenging listening conditions. Next, I discuss whether intelligibility-enhancing speaking style modifications impact speech processing beyond word recognition, namely recognition memory for sentences. The results show that effortful speech processing in challenging listening environments can be improved by speaking style adaptations on the part of the talker. In addition to enhanced intelligibility, a substantial improvement in sentence recognition memory can be achieved through speaker adaptations to the environment and to the listener when in adverse conditions. These results have implications for the quality of speech communication in a variety of environments, such as classrooms and hospitals.

9:25

4aAAa3. Reducing cognitive demands on listeners by speaking clearly in noisy places. Kristin Van Engen (Psych., Washington Univ. in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899, kvanengen@wustl.edu)

Listeners have more difficulty identifying spoken words in noisy environments when those words have many phonological neighbors (i.e., similar-sounding words in the lexicon) than when they have few phonological neighbors. This difficulty appears to be exacerbated in old age, where reductions in inhibitory control presumably make it more difficult to cope with competition from similar-sounding words. Fortunately, word recognition in noise can generally be improved for a wide range of listeners (e.g., younger and older adults, individuals with and without hearing impairment) when speakers adopt a clear speaking style. This study investigated whether clear speech, in addition to generally increasing speech intelligibility, also reduces the inhibitory demands associated with identifying lexically difficult words in noise for younger and older adults. The results show that, indeed, the difference between rates of identification for words with many versus few neighbors was eliminated when those words were produced in clear speech. Data on the roles of individual differences (e.g., hearing, working memory, and inhibitory control) that may contribute to word identification in noise will also be presented.

9:45

4aAAa4. Improved speech understanding and amplitude modulation sensitivity in rooms: Wait a second!. Pavel Zahorik (Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville School of Medicine, Psychol. and Brain Sci., Life Sci. Bldg. 317, Louisville, KY 40292, pavel.zahorik@louisville.edu), Paul W. Anderson (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), Eugene Brandewie (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), and Nirmal K. Srinivasan (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., Portland, OR)

Sound transmission between source and receiver can be profoundly affected by room acoustics, yet under many circumstances, these acoustical effects have relatively minor perceptual consequences. This may be explained, in part, by listener adaptation to the acoustics of the listening environment. Here, evidence that room adaptation improves speech understanding is summarized. The adaptation is rapid (around 1 s), and observable for a variety of speech materials. It also appears to depend critically on the amplitude modulation characteristic of the signal reaching the ear, and as a result, similar room adaptation effects have been observed for measurements of amplitude modulation sensitivity. A better understanding of room adaptation effects will hopefully contribute to improved methods for speech transmission in rooms for both normally hearing and hearing-impaired listeners. [Work supported by NIDCD.]

10:05–10:20 Break

10:20

4aAAa5. The importance of attention, localization, and source separation to speech cognition and recall. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt Auburn St #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Acoustic standards for speech are based on word recognition. But for successful communication sound must be detected, separated from noise and other streams, phones, syllables, and words must be recognized and parsed into sentences, meaning must be found by relating the sentences to previous knowledge, and finally information must be stored in long term memory. All of these tasks require time and working memory. Acoustical conditions that increases the difficulty of any part of the task reduce recall. But attention is possibly the most important factor in successful communication. There is compelling anecdotal evidence that sound profoundly and involuntarily influences attention. Humans detect in fractions of a second whether a sound source is close, independent of its loudness and frequency content. When sound is perceived as close it demands a degree of attention that distant sound does not. The mechanism of detection relies on the phase relationships between harmonics of complex tones in the vocal formant range, properties of sound that also ease word recognition and source separation. We will present the physics of this process and the acoustic properties that enable it. Our goal is to increase attention and recall in venues of all types.

10:40

4aAAa6. Release from masking in simulated reverberant environments. Nirmal Kumar Srinivasan, Frederick J. Gallun, Sean D. Kampel, Kasey M. Jakien, Samuel Gordon, and Megan Stansell (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 in understanding speech in complex listening environments. In two separate experiments, speech intelligibility enhancement due to prior exposure to listening environment and spatial release from masking (SRM) for small spatial separations were measured in simulated reverberant listening environments. Release from masking was measured by comparing threshold target-to-masker ratios (TMR) obtained with a speech target presented directly ahead of the listener and two speech maskers presented from the same location or in symmetrically displaced spatial configurations in an anechoic chamber. The results indicated that older listeners required much higher TMR at threshold and obtained decreased benefit from prior exposure to listening environments compared to younger listeners. For the small separation experiment, speech stimuli were presented over headphones and virtual acoustic techniques were used to simulate very small spatial separations (approx. 2 degrees) between target and maskers. Results reveal, for the first time, the minimum separation required between target and masker to achieve release from speech-on-speech masking in anechoic and reverberant conditions. The advantages of including small separations for understanding the functions relating spatial separation to release from masking will be discussed, as well as the value of including older listeners. [Work supported by NIH R01 DC011828.]

11:00

4aAAa7. Speech-on-speech masking for children and adults. Lauren Calandruccio, Lori J. Leibold (Allied Health Sci., Univ. of North Carolina, 301 S. Columbia St., Chapel Hill, NC 27599, Lauren_Calandruccio@med.unc.edu), and Emily Buss (Otolaryngology/Head and Neck Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Children experience greater difficulty understanding speech in noise compared to adults. This age effect is pronounced when the noise causes both energetic and informational masking, for example, when listening to speech while other people are talking. As children acquire speech and language, they are faced with multi-speech environments all the time, for example, in the classroom. For adults, speech perception tends to be worse when the target and masker are matched in terms of talker sex and language, with mismatches improving performance. It is unknown, however, whether children are able to benefit from these (sex or language) target/masker mismatches. The goal of this project is to further our understanding of the speech-on-speech masking deficit children demonstrate throughout childhood, while specifically investigating whether children's speech recognition improves when the target and masker are spoken by talkers of the opposite sex, or when the target and masker speech are spoken in different languages. Normal-hearing children and adults were tested on word identification and sentence recognition tasks. Differences in SNR needed to equate performance between the two groups will be reported, as well as data reporting whether children are able to benefit from these target/masker mismatch cues.

11:20

4aAAa8. The neural basis of informational and energetic masking effects in the perception and production of speech. Samuel Evans (Inst. of Cognit. Neurosci., Univ. College London, 17 Queen Square, London, London WC1N 3AR, United Kingdom, samuel.evans@ucl.ac.uk), Carolyn McGettigan (Dept. of Psych., Royal Holloway, Egham, United Kingdom), Zarinah Agnew (Dept. of Otolaryngol., Univ. of California, San Francisco, San Francisco, CA), Stuart Rosen (Dept. of Speech, Hearing and Phonetic Sci., Univ. College London, London, United Kingdom), Lima Cesar (Ctr. for Psych., Univ. of Porto, Porto, Portugal), Dana Boebinger, Markus Ostarek, Sinead H. Chen, Angela Richards, Sophie Meekings, and Sophie K. Scott (Inst. of Cognit. Neurosci., Univ. College London, London, United Kingdom)

When we have spoken conversations, it is usually in the context of competing sounds within our environment. Speech can be masked by many different kinds of sounds, for example, machinery noise and the speech of others, and these different sounds place differing demands on cognitive resources. In this talk, I will present data from a series of functional magnetic resonance imaging (fMRI) studies in which the informational properties of background sounds have been manipulated to make them more or less similar to speech. I will demonstrate the neural effects associated with speaking over and listening to these sounds, and demonstrate how in perception these effects are modulated by the age of the listener. The results will be interpreted within a framework of auditory processing developed from primate neurophysiology and human functional imaging work (Rauschecker and Scott 2009).

THURSDAY MORNING, 30 OCTOBER 2014 — SANTA FE, 10:35 A.M. TO 12:05 P.M.

Session 4aAAb

Architectural Acoustics: Uses, Measurements, and Advancements in the Use of Diffusion and Scattering Devices

David T. Bradley, Chair
Physics Astronomy, Vassar College, Poughkeepsie, NY 12604

Chair's Introduction—10:35

Invited Papers

10:40

4aAAb1. Effect of installed diffusers on sound field diffusivity in a real-world classroom. Ariana Sharma, David T. Bradley, and Mohammed Abdelaziz (Phys. + Astronomy, Vassar College, 124 Raymond Ave, Poughkeepsie, NY 12604, arsharma@vassar.edu)

An ideal diffuse sound field is both homogeneous (acoustic quantities are independent of position) and isotropic (acoustic quantities are invariant with respect to direction). Predicting and characterizing sound field diffusivity is essential to acousticians when designing and using acoustically sensitive spaces. Surfaces with a non-planar geometry, referred to as diffusers, can be installed in these spaces as a means of increasing and/or controlling the field diffusivity. Although some theoretical and computational modeling work has been carried out to better understand the relationship between these installed diffusers and the resulting field diffusivity, the current state-of-the-art does not include a systematic understanding of this relationship. Furthermore, very little work has been done to characterize this relationship in full scale and in the real world. In the current project, the effect of diffusers on field diffusivity has been studied in a full scale, real-world classroom. Field diffusivity has been measured for various configurations of the diffusers using two measurement techniques. The first technique uses a three-dimensional grid of receivers to characterize the field homogeneity. To characterize field isotropy, a spherical microphone array has also been used. Results and analysis will be presented and discussed.

11:00

4aAAb2. Effect of measurement conditions on sound scattered from a pyramid diffuser in a free field. Kimberly A. Riegel, David T. Bradley, Mallory Morgan, and Ian Kowalok (Phys. + Astronomy, Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604, kiriegel@vassar.edu)

A surface with a non-planar geometry, referred to as a diffuser, can be used in acoustically sensitive spaces to help control or eliminate unwanted effects from strong reflections by scattering the reflected sound. The scattering behavior of a diffuser can be measured in a free field, according to the standard ISO 17497-2. Many of the measurement conditions discussed in this standard can have an effect on the measured data; however, these conditions are often not well-specified and/or have not been substantiated. In the current study, a simple pyramid diffuser has been measured while varying several measurement conditions: surface material, orientation of the surface geometry, perimeter shape of the surface, and mounting depth of the surface. Reflected polar response and diffusion coefficient data have been collected and compared for each condition. Data have also been contrasted with those obtained by numerical simulation using boundary element method (BEM) techniques for an idealized pyramid diffuser. Results and analysis will be presented and discussed.

Contributed Papers

11:20

4aAAb3. Sound field diffusion by number of peak by continuous wavelet transform. Yongwon Cha, Muhammad Imran, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Hanyang University, Seoul 133-791, South Korea, chadyongwoncha@gmail.com)

The number of peak (Np) in the impulse response signal (IRs) captured for the real hall have been investigated and measured by using continuous wavelet transform (CWT). Np has a relationship with perceptual diffusion as an objective characteristic that is influenced by walls scattering elements. In addition, as measuring diffuse sound fields, the CWT coefficients are used for detecting the diffusive sound. Based on the absolute coefficient values calculated from CWT analysis, a practical method of counting reflections is considered. These reflections are specified as diffusive or specular base on their similarity with the mother wavelet. Temporal and spatial representation of absolute values of CWT is presented. Auditory experiments using a paired comparison method were conducted to gauge the relationship between the Np and perceptual sound field diffusion. It is revealed that a dominant factor influencing the subjective preference in the hall was the Np that varied with different wall surface treatments.

11:35

4aAAb4. In praise of smooth surfaces: Promoting a balance between specular and diffuse surfaces in performance space design. Gregory A. Miller and Scott D. Pfeiffer (Threshold Acoust., LLC, 53 W. Jackson Boulevard, Ste. 815, Chicago, IL 60604, gmiller@thresholdacoustics.com)

Diffusive surfaces are often presented as a panacea for achieving desirable listening conditions in performance spaces. While diffusive surfaces are a valuable and necessary part of the finish palette in any theater or concert hall, a significant number of specular surfaces are crucial to the success of many such spaces. Case studies will be presented in which excessive use of diffusion has resulted in losses of clarity and loudness, including comparisons to the results following the introduction of specular surfaces, either flat or gently curved. Aural examples will be presented to demonstrate the perceptual differences when specular surfaces are employed as compared to highly diffusive surfaces at key locations in spaces for music and drama.

11:50

4aAAb5. Scattershot: A look at designing, integrating, and measuring diffusion. Shane J. Kanter, John Strong, Carl Giegold, and Scott Pfeiffer (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com)

A primary goal of the small-scale performance venue is to provide the audience with supportive, well-timed reflections and to energize the space adequately without overpowering the room volume. The judicious use of sound-diffusive elements in such venues can lend a pleasing sense of body and space while avoiding undesirable reflections that disrupt the listener experience. However, while working with architects to develop a space that is pleasing to both the ear and the eye, it is often necessary to reconcile these needs with each other. Diffusive elements must integrate seamlessly within the space visually as well as architecturally. While developing interior room acoustics for three small spaces for performance/worship, with audience size ranging from 150 to 299, an exploration of diffusive elements was conducted. As each project required a different method and frequency range of diffusion, scale models were constructed and tested under varied conditions, using sometimes unorthodox methods to determine the acoustic effect. These efforts were focused on limiting coloration caused by the "picket fence effect," reducing harsh reflections without rendering a space excessively sound-absorptive, and maintaining coherent reflections from discrete sections of a prominent wall while leaving other sections diffusive. Methods, experiences, and results will be presented.

Session 4aAB

Animal Bioacoustics and Acoustical Oceanography: Use of Passive Acoustics for Estimation of Animal Population Density I

Tina M. Yack, Cochair
Bio-Waves, Inc., 364 2nd Street, Suite #3, Encinitas, CA 92024

Danielle Harris, Cochair
Centre for Research into Ecological and Environmental Modelling, University of St. Andrews, The Observatory, Buchanan Gardens, St. Andrews KY16 9LZ, United Kingdom

Chair's Introduction—8:00

Invited Papers

8:05

4aAB1. Estimating density from passive acoustics: Are we there yet? Tiago A. Marques, Danielle Harris, and Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, The Observatory, Buchannan Gardens, St. Andrews, Fife KY16 9 LZ, United Kingdom, tiago.marques@st-andrews.ac.uk)

In the last few years, there have been a considerable number of papers describing methods or case studies involving passive acoustic density estimation. While this might be interpreted as evidence that density estimates might now be easily and routinely implemented, the truth is that so far these methods and applications have been essentially proof-of-concept in nature, based in areas and/or species particularly suited for the methods and also often involved assumptions hard to evaluate. We briefly review some of the existing work in this area concentrating on a few aspects we believe are key for the implementation of density estimation from passive acoustics in a broader context. These are (1) the development of fundamental research addressing the problem of sound production rate, fundamental as it allows to convert estimates of density of sounds into density of animals and (2) the development of hardware capable of providing cheap deployable units capable of ranging, allowing straightforward implementations of distance sampling based approaches. The perfect density estimate is out there waiting to happen, but we have not found it yet.

8:25

4aAB2. Use of passive acoustics for estimation of cetacean population density: Realizing the potential. Jay Barlow and Shannon Rankin (Marine Mammal and Turtle Div., NOAA-SWFSC, 8901 La Jolla Shores Dr., La Jolla, CA 92037, jay.barlow@noaa.gov)

The potential of passive acoustic methods to estimate cetacean population density has seldom been realized. It has been most successfully applied to species that consistently use echo-location during foraging, have very distinctive echo-location signals and forage a large fraction of the time, notably sperm whale, porpoise, and beaked whales. Research is needed to eliminate some of the impediments to applying acoustics to estimate the density of other species. For baleen whales, one of the greatest uncertainties is the lack of information on call rates. For delphinids, the greatest uncertainties are in estimating group size and in species recognition. For all species, there is a need to develop inexpensive recorders that can be distributed in large number at random locations in a study area. For towed hydrophone surveys, there is a need to better localize species in their 3-D environment and to instantaneously localize animals from a single signal received on multiple hydrophones. While improvements can be made, we may need to recognize that some of impediments cannot be overcome with any reasonable research budget. In these cases, efforts should be concentrated in improving acoustic methods to aid visual-based transect methods.

8:45

4aAB3. Acoustic capture-recapture methods for animal density estimation. David Borchers (Dept. of Mathematics & Statistics, Univ. of St. Andrews, CREEM, Buchannan Gdns, St. Andrews, Fife KY16 9LZ, United Kingdom, dlb@st-andrews.ac.uk)

Capture-recapture methods are one of the two most widely-used methods of estimating wildlife density and abundance. They can be used with passive acoustic detectors—in which case acoustic detection on a detector constitutes "capture" and detection on other detectors and/or at other times constitute "recaptures." Unbiased estimation of animal density from any capture-recapture survey requires that the effective area of the detectors be estimated, and information on detected animals' locations are essential for this. While locations are not observed, acoustic data contain information on location in a variety of guises, including time-difference-of arrival, signal strength, and sometimes directional information. This talk gives an overview of the use of such data with spatially explicit capture-recapture (SECR) methods, including consideration of some of the particular challenges that acoustic data present for SECR methods, ways of dealing with these, and an outline of some unresolved issues.

9:05

4aAB4. U.S. Navy application and interest in passive acoustics for estimation of marine mammal population density. Anu Kumar (Living Marine Resources, NAVFAC EXWC, 1000 23rd Ave., Code EV, Port Hueneme, CA 93043, anurag.kumar@navy.mil), Chip Johnson (Environ. Readiness, Command Pacific Fleet, Coronado, CA), Julie Rivers (Environ. Readiness, Command Pacific Fleet, Pearl Harbor, HI), Jene Nissen (Environ. Readiness, U.S. Fleet Forces, Norfolk, VA), and Joel Bell (Marine Resources, NAVFAC Atlantic, Norfolk, VA)

Marine species population density estimation from passive acoustic monitoring is an emergent topic of interest to the U.S. Navy. Density estimates are used by the Navy and other Federal partners in effects modeling for environmental compliance documentation. Current traditional methods of marine mammal density estimation via visual line transect surveys require expensive ship time and long days at-sea for an experienced crew to yield limited spatial and temporal coverage. While visual surveys remain an effective means of deriving density estimates, passive acoustic based density estimation methods have the unique ability to improve on visual density estimates for some key species by: (a) expanding spatial and temporal density coverage, (b) providing coverage in areas too remote or difficult for traditional visual surveys, (c) reduce the statistical uncertainty of a given density estimate, and (d) providing estimates for species that are difficult to survey visually (e.g., minke and beaked whales). The U.S. Navy has invested in research for the development, refinement, and scientific validation of passive acoustic methods for cost effective density estimates in the future. The value, importance, and current development in passive acoustic-based density estimation methods for Navy applications will be discussed.

9:25

4aAB5. Towing the line: Line-transect based density estimation of whales using towed hydrophone arrays. Thomas F. Norris and Tina M. Yack (Bio-Waves Inc., 364 2nd St., Ste. #3, Encinitas, CA 92024, thomas.f.norris@bio-waves.net)

Towed hydrophone arrays have been used to monitor marine mammals from research vessels since the 1980s. Although towed hydrophone arrays have now become a standard part of line-transect surveys of cetaceans, density estimation exclusively using passive acoustic has only been attempted for a few species. We use examples from four acoustic line-transect surveys that we conducted in the North Pacific Ocean to illustrate the steps involved, and issues inherent, in using data from towed hydrophone arrays to estimate densities of cetaceans. We will focus on two species of cetaceans, sperm whales and minke whales, with examples of beaked whales and other species as needed. Issues related to survey design, data-collection, and data analysis and interpretation will be discussed using examples from these studies. We provide recommendations to improve the survey design, data-collection methods, and analyses. We also suggest areas where additional research and methodological development are required in order to produce robust density estimates from acoustic based data.

Contributed Papers

9:45

4aAB6. From clicks to counts: Applying line-transect methods to passive acoustic monitoring of sperm whales in the Gulf of Alaska. Tina M. Yack, Thomas F. Norris, Elizabeth Ferguson (Bio-Waves Inc., 364 2nd St., Ste. #3, Encinitas, CA 92024, tina.yack@bio-waves.net), Brenda K. Rone (Cascadia Res. Collective, Seattle, WA), and Alexandre N. Zerbini (Alaska Fisheries Sci. Ctr., Seattle, WA)

A visual and acoustic line-transect survey of marine mammals was conducted in the central Gulf of Alaska (GoA) during the summer of 2013. The survey area was divided into four sub-strata to reflect four distinct habitats; "inshore," "slope," "offshore," and "seamount." Passive acoustic monitoring was conducted using a towed-hydrophone array system. One of the main objectives of the acoustic survey was to obtain an acoustic-based density estimate for sperm whales. A total of 241 acoustic encounters of sperm whales during 6,304 km of effort were obtained compared to 19 visual encounters during 4,155 km of effort. Line-transect analytical methods were used to estimate the abundance of sperm whales. To estimate the detection function, target motion analysis was used to obtain perpendicular distances to individual sperm whales. An acoustic-based density and abundance estimate was obtained for each stratum ($N = 78$; CV = 0.36 offshore; $N = 16$; CV = 0.55 seamount; $N = 121$; and CV = 0.18 slope) and for the entire survey area ($N = 215$; $D = 0.0013$; and CV = 0.18). These results will be compared to visual-based estimates. The advantages and disadvantages of acoustic-based density estimates as well as application of these methods to other species (e.g., beaked whales) and areas will be discussed.

10:00–10:15 Break

10:15

4aAB7. Studying the biosonar activities of deep diving odontocetes in Hawaii and other western Pacific locations. Whitlow W. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu) and Giacomo Giorli (Oceanogr. Dept., Univ. of Hawaii, Honolulu, HI)

Ecological acoustic recorders (EARs) have been deployed at several locations in Hawaii and in other western Pacific locations to study the foraging behavior of deep-diving odontocetes. EARs have been deployed at depths greater than 400 m at five locations around the island of Kauai, one at Ni'ihau, two around the island of Okinawa, and four in the Marianas (two close to island of Guam, one close to the island of Saipan and another close to the island of Tinian). The four groups of deep-diving odontocetes were blackfish (mainly pilot whales and false killer whales), sperm whales, beaked whales (Cuvier and Bainsville beaked whales) and Risso's dolphin. In all locations, the biosonar signals of blackfish were detected the most followed by either by sperm and beaked whales depending on specific locations with Risso's dolphin being detected the least. There was a strong tendency for these animals to forage at night in all locations. The detection rate indicate much lower populations of these four groups of odontocetes around Okinawa and in the Marianas then off Kauai in the main Hawaiian island chain by a factor of about 4–5.

10:30

4aAB8. Fin whale vocalization classification and abundance estimation. Wei Huang, Delin Wang (Elec. and Comput. Eng., Northeastern Univ., 006 Hayden Hall, 370 Huntington Ave., Boston, MA 02115, huang.wei1@husky.neu.edu), Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Several thousand fin whale vocalizations from multiple fin individuals were passively recorded by a high-resolution coherent hydrophone array system in the Gulf of Maine in Fall 2006. The recorded fin whale vocalizations have relatively short durations roughly 0.4 s and frequencies ranging

from 15 to 40 Hz. Here we classify the fin whale vocalizations and apply the results to estimate the minimum number of vocalizing fin individuals detected by our hydrophone array. The horizontal azimuth or bearing of each fin whale vocalization is first determined by beamforming. Each beamformed fin whale vocalization spectrogram is next characterized by several features such as center frequency, upper and lower frequency limits, as well as amplitude-weighted mean frequency. The vocalizations are then classified using k-mean clustering into several distinct vocal types. The vocalization clustering result is then combined with the bearing-time trajectory information for a consecutive sequence of vocalizations to provide an estimate of the minimum number of vocalizing fin individuals detected.

10:45

4aAB9. Neglect of bandwidth of odontocetes echolocation clicks biases propagation loss and single hydrophone population estimates. Michael A. Ainslie, Alexander M. von Benda-Beckmann (Acoust. and Sonar, TNO, P.O. Box 96864, The Hague 2509JG, Netherlands, michael.ainslie@tno.nl), Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St Andrews, United Kingdom), and Tyack L. Tyack (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, United Kingdom)

Passive acoustic monitoring with a single hydrophone has been suggested as a cost-effective method to monitor population density of echolocating marine mammals, by estimating the distance at which the hydrophone is able to distinguish the echolocation clicks from the background. To avoid a bias in the estimated population density, this method relies on an unbiased estimate of the propagation loss (PL). It is common practice to estimate PL at the center frequency of a broadband echolocation click and to assume this narrowband PL applies also to the broadband click. For a typical situation this narrowband approximation overestimates PL, underestimates the detection range and consequently overestimates the population density by an amount that for fixed center frequency increases with increasing pulse bandwidth and sonar figure of merit. We investigate the detection process for different marine mammal species and assess the magnitude of error on the estimated density due to various simplifying assumptions. Our main purposes are to quantify and, where possible and needed, correct the bias in the population density estimate for selected species and detectors due to use of the narrowband approximation, and to understand the factors affecting the magnitude of this bias to enable extrapolation to other species and detectors.

11:00

4aAB10. Instantaneous acoustical response of marine mammals to abrupt changes in ambient noise. John E. Joseph, Tetyana Margolina (Oceanogr., Naval Postgrad. School, 833 Dyer Rd, Monterey, CA, jejoseph@nps.edu), and Ming-Jer Huang (National Kaohsiung Univ. of Appl. Sci., Kaohsiung, Taiwan)

Four months of passive acoustic data recorded at Thirtymile Bank in offshore southern California have been analyzed to describe instantaneous vocal response of marine mammals to abrupt changes in ambient noise. Main contributors to the distinctive regional soundscape are heavy commercial shipping, military activities in the naval training range, diverse marine life and natural sources including wind and tectonic activity. Many of these sources produce intense, irregular and short-term events shaped by local oceanographic conditions, bathymetry and bottom structure (Thirtymile Bank blind thrust). We seek to attribute detected changes in cetacean vocal behavior (loudness, calling rate, and pattern) to these events and differentiate the reaction by noise source, its intensity, frequency and/or duration. Main target species are blue and fin whales. Initial hypotheses formulated after data scanning are tested statistically (2D histograms and PCA). To quantify the vocal behavior variations, an innovative detection approach based on pattern recognition is applied, which allows for extraction of individual calls with low false alarm and high detection success comparable to those of a human analyst. Obtained results relate cetacean acoustic behavior to ambient noise variability and thus help refine existing cue-based formulae for estimation of whale population density from PAM data.

11:15

4aAB11. Measuring whale and dolphin call rates as a function of behavioral, social, and environmental context. Stacy L. DeRuiter, Catriona M. Harris (School of Mathematics & Statistics, Univ. of St. Andrews, CREEM, St. Andrews KY169LZ, United Kingdom, sldr@st-andrews.ac.uk), Nicola J. Quick (Duke University Marine Lab, Duke Univ., Beaufort, NC), Dina Sadykova, Lindesay A. Scott-Hayward (School of Mathematics & Statistics, Univ. of St. Andrews, St. Andrews, United Kingdom), Alison K. Stimpert (Moss Landing Marine Lab., California State Univ., Moss Landing, CA), Brandon L. Southall (Southall Environ. Assoc., Inc., Aptos, CA), Len Thomas (School of Mathematics & Statistics, Univ. of St. Andrews, St Andrews, United Kingdom), and Fleur Visser (Kelp Marine Res., Hoorn, Netherlands)

Cetacean sound-production rates are highly variable and patchy in time, depending upon individual behavior, social context, and environmental context. Better quantification of the drivers of this variability should allow more realistic estimates of expected call rates, improving our ability to convert between call counts and animal density, and also facilitating detection of sound-production changes due to acoustic disturbance. Here, we analyze digital acoustic tag (DTAG) records and visual observations collected during behavioral response studies (BRSs), which aim to assess normal cetacean behavior and measure changes in response to acoustic disturbance; data sources include SOCAL BRS, the 3S project, and Bahamas BRS, with statistical contributions from the MOCHA project (http://www.creem.st-and.ac.uk/mocha/links). We illustrate use of generalized linear models (and their extensions) as a flexible framework for sound-production-rate analysis. In the context of acoustic disturbance, we also detail use of two-dimensional spatially adaptive surfaces to jointly model effects of sound-source proximity and sound intensity. Specifically, we quantify variability in pilot whale group sound production rates in relation to behavior and environment, and individual fin whale call rates in relation to social and environmental context and dive behavior; with and without acoustic disturbance.

11:30

4aAB12. Estimating relative abundance of singing humpback whales in Los Cabos, México, using diffuse ambient noise. Kerri Seger, Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8880 Biological Grade, MESOM 161, La Jolla, CA 92093-0206, kseger@ucsd.edu), Diana C. López Arzate, and Jorge Urbán (Laboratorio de Mamíferos Marinos, Universidad Autónoma de Baja California Sur, La Paz, BCS, Mexico)

Previous research has speculated that diffuse ambient noise levels can be used to estimate relative cetacean abundance in certain locations when baleen whale vocal activity dominates the soundscape (Au *et al*., 2000; Mellinger *et al*., 2009). During the 2013 and 2014 humpback whale breeding seasons off Los Cabos, Mexico, visual point and line transects were conducted alongside two bottom-mounted acoustic deployments. As theorized, preliminary analysis of ambient noise between 100 and 1,000 Hz is dominated by humpback whale song. It also displays a diel cycle similar to that found in the West Indies, Australia, and Hawai'i, whereby peak levels occur near midnight and troughs occur soon after sunrise (Au *et al*., 2000; McCauley *et al*., 1996). Depending upon site and year, the median band-integrated levels fluctuated between 7 and 16 dB re 1 uPa when sampled in one hour increments. This presentation uses analytical models of wind-generated noise in an ocean waveguide to analyze potential relationships between singing whale density and diffuse ambient noise levels. It explores whether various diel cycle strengths (peak-to-peak measurements and Fourier analysis) correspond with trends observed from concurrent visual censuses. [Work sponsored by the Ocean Foundation.]

11:45

4aAB13. Large-scale static acoustic survey of a low-density population—Estimating the abundance of the Baltic Sea harbor porpoise. Jens C. Koblitz (German Oceanogr. Museum, Katharinenberg 14-20, Stralsund 18439, Germany, Jens.Koblitz@meeresmuseum.de), Mats Amundin (Kolmården Wildlife Park, Kolmarden, Sweden), Julia Carlström (AquaBiota Water Res., Stockholm, Sweden), Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), Ida Carlén (AquaBiota Water Res., Stockholm, Sweden), Jonas Teilmann (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Nick Tregenza (Chelonia Ltd., Long Rock, United Kingdom), Daniel Wennerberg (Kolmården Wildlife Park, Kolmarden, Sweden), Line Kyhn, Signe Svegaard (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Radek Koza, Monika Kosecka, Iwona Pawliczka (Univ. of Gdansk, Gdansk, Poland), Cinthia Tiberi Ljungqvist (Kolmården Wildlife Park, Kolmården, Sweden), Katharina Brundiers (German Oceanogr. Museum, Stralsund, Germany), Andrew Wright (George Mason Univ., Fairfax, VA), Lonnie Mikkelsen, Jakob Tougaard (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Olli Loisa (Turku Univ. of Appl. Sci., Turku, Finland), Anders Galatius (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Ivar Jüssi (ProMare NPO, Harjumaa, Estonia), and Harald Benke (German Oceanogr. Museum, Stralsund, Germany)

SAMBAH (Static Acoustic Monitoring of the Baltic Sea Harbor Porpoise) is an EU LIFE+-funded project with the primary goal of estimating the abundance and distribution of the critically endangered Baltic Sea harbor porpoise. From May 2011 to April 2013, project members in all EU countries around the Baltic Sea undertook a static acoustic survey using 304 porpoise detectors distributed in a randomly positioned systematic grid in waters 5–80 m deep. In the recorded data, click trains originating from porpoises have been identified automatically using an algorithm developed specifically for Baltic conditions. To determine the click train C-POD detection function, a series of experiments have been carried out, including acoustic tracking of wild free ranging porpoises using hydrophone arrays in an area with moored C-PODs and playbacks of porpoise-like signals at SAMBAH C-PODs during various hydrological conditions. Porpoise abundance has been estimated by counting the number of individuals detected in short time interval windows (snapshots), and then accounting for false positive detections, probability of animals being silent, and probability of detection of non-silent animals within a specified maximum range. We describe the method in detail, and how the auxiliary experiments have enabled us to estimate the required quantities.

THURSDAY MORNING, 30 OCTOBER 2014 INDIANA A/B, 7:55 A.M. TO 12:00 NOON

Session 4aBA

Biomedical Acoustics: Mechanical Tissue Fractionation by Ultrasound: Methods, Tissue Effects, and Clinical Applications I

Vera A. Khokhlova, Cochair
University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Jeffrey B. Fowlkes, Cochair
Univ. of Michigan Health System, 3226C Medical Sciences Building I, 1301 Catherine Street, Ann Arbor, MI 48109-5667

Chair's Introduction—7:55

Invited Papers

8:00

4aBA1. Histotripsy: An overview. Charles A. Cain (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., 2121 Gerstacker, Ann Arbor, MI 48105, cain@umich.edu)

Histotripsy produces non-thermal lesions by generating dense highly confined energetic bubble clouds that mechanically fractionate tissue. This nonlinear thresholding phenomenon has useful consequences. If only the tip of the waveform (P-) exceeds the intrinsic threshold*, small lesions less than the diffraction limit can be generated. This is called microtripsy (other presentations in this session). Moreover, side lobes from distorting aberrations can be "thresholded-out" wherein part of the main lobe exceeds the intrinsic threshold producing a clean bubble cloud (and lesion) conferring significant immunity to aberrations. If a high frequency probe (imaging) waveform intersects a low frequency pump waveform, the compounded waveform can momentarily exceed the intrinsic threshold producing a lesion with an imaging transducer. Multi-beam histotripsy (other presentations in this session) allows flexible placement of both pump and probe transducers. Very broadband P- "monopolar" pulses*, ideal for histotripsy, can be synthesized in a generalization of the multi-beam histotripsy (other presentations in this session) case wherein very short pulses from transducer elements of many different frequencies are added at the focus of what is called a frequency compounding transducer (other presentations in this session). Ultrasound image guidance works well with histotripsy. Bubble clouds are easily seen simplifying both lesion targeting and continuous validation of the ongoing process. Hypoechoic homogenized tissue allows real-time quantification of lesion formation.

8:20

4aBA2. Boiling histotripsy: A noninvasive method for mechanical tissue disintegration. Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, 1013 NE 40th St., Seattle, WA 98105, amax38@u.washington.edu), Tatiana D. Khokhlova (Dept. of Gastroenterology, Univ. of Washington, Seattle, WA), George R. Schade (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Yak-Nam Wang, Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Petr Yuldashev (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Julianna C. Simon (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Navid Farr (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Ari Partanen (Clinical Sci., Philips Healthcare, Cleveland, OH), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Joo Ha Hwang (Dept. of Gastroenterology, Univ. of Washington, Seattle, WA), Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Boiling histotripsy is an experimental noninvasive focused ultrasound therapy that applies shocked ms-length pulses to achieve mechanical disintegration of a targeted tissue. Localized delivery of high-amplitude shocks causes rapid heating, resulting in boiling of the tissue. The interaction of incident shocks with the boiling bubble results in tissue disruption and liquefaction without significant thermal injury. Simulations are utilized to design and characterize therapy sources, predicting focal waveforms, shock amplitudes, and boiling times. Transducers have been developed to generate focal shock amplitudes >70 MPa and achieve rapid boiling at depth in tissue. Therapy systems including ultrasound-guided single-element sources and clinical MRI-guided phased arrays have been successfully used to create *ex vivo* and *in vivo* lesions at ultrasound frequencies in the 1–3 MHz range. Histological and biochemical analyses show mechanical disruption of tissue architecture with minimal thermal effect, similar to cavitation-based histotripsy. Atomization as observed with acoustic fountains has been proposed as an underlying mechanism of tissue disintegration. This promising technology is being explored for several applications in tissue ablation, as well as new areas such as tissue engineering and biomarker detection. [Work supported by NIH 2T32DK007779-11A1, R01EB007643-05, 1K01EB015745, and NSBRI through NASA NCC 9-58.]

8:40

4aBA3. Bubbles in tissue: Yes or No? Charles C. Church (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, cchurch@olemiss.edu)

The question of whether bubbles exist in most or all biological tissues rather than being restricted to only a few well-known examples remains a mystery. When Apfel and Holland developed the theoretical background for the mechanical index (MI), they first assumed that such bubbles did exist and further assumed that some of those bubbles were of a size that would undergo inertial cavitation at the lowest possible rarefactional pressure. Comparison of cavitation thresholds determined experimentally in various mammalian tissues *in vivo* with the results of computational studies seems to provide a definitive answer to that question. No, optimally sized bubbles do not pre-exist in tissue, although very small bubbles, with radii on the order of nm, may be present. However, this answer is inextricably related to the accuracy of the theory used to study the question, in this case a form of the Keller-Miksis equation modified to include the viscoelastic properties of tissue. Previous analysis has focused on elasticity, assuming that viscosity is constant, but is it? Blood is known to be shear-thinning, and some soft tissues appear to be as well. The effect of shear rate on cavitation thresholds and implications for bubble populations in tissue will be discussed.

9:00

4aBA4. Benefits and challenges of employing elevated acoustic output in diagnostic imaging. Kathryn Nightingale (Biomedical Eng., Duke Univ., PO Box 90281, Durham, NC 27708-0281, kathy.nightingale@duke.edu) and Charles C. Church (National Ctr. for Acoust., Univ. of MS, University, MS)

The acoustic output levels used in diagnostic ultrasonic imaging in the US have been subject to a de facto limitation by guidelines established by the USFDA in 1976, for which no known bioeffects had been reported. These track-3 guidelines link the Mechanical Index (MI) and the Thermal Index (TI) to the maximum outputs as of May 28, 1976, through a linear derating process. Subsequently, new imaging technologies have been developed that employ unique beam sequences (e.g., harmonic imaging and ARFI imaging) which were not well developed when the current regulatory scheme was put in place, so neither the MI nor the TI takes them into account in an optimal manner. Additionally, there appears to be a large separation between the maxima in the track-3 guidelines and the acoustic output levels for which cavitation-based bioeffects are observed in tissues not known to contain gas bodies. In this presentation, we summarize the history of and the scientific basis for the MI, define an output regime and specify clinical applications under consideration for conditionally increased output (CIO), review the potential risks of CIO in this regime based upon existing scientific evidence, and summarize the evidence for the potential clinical benefits of CIO.

9:20

4aBA5. Standards for characterizing highly nonlinear acoustic output from therapeutic ultrasound devices: Current methods and future challenges. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

One of the major challenges of characterizing the acoustic fields and power from diagnostic and high intensity or high pressure therapeutic devices is addressing the impact of amplitude-dependent nonlinear propagation effects. The destructive capabilities of high intensity therapeutic devices (HITU) make acoustic output measurements with conventional fragile sensors used for diagnostic ultrasound difficult. Different approaches involving more robust measurement devices, scaling and simulation are described in two recent IEC documents, IEC TS 62556 for the specification and measurement of HITU fields and IEC 62555 for the measurement of acoustic power from HITU devices. Existing and proposed applications include even higher pressure levels and use of cavitation effects. Promising hybrid approaches involve a combination of measurement and simulation. In order to meet the challenges of design, verification, and measurement, standards and consensus are needed to couple the measurements to the prediction of acoustic output in realistic tissue models as well as associated effects such as acoustic radiation force and temperature elevation.

9:40

4aBA6. Uncertainties in characterization of high-intensity, nonlinear pressure fields for therapeutic applications. Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Petr V. Yuldashev (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Tatiana D. Khokhlova (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Sergey A. Tsysar (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Michael R. Bailey (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov, and Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

A fundamental aspect of developing therapeutic ultrasound applications is the need to quantitatively characterize the acoustic fields delivered by transducers. A typical approach is to make direct pressure measurements in water. With very high intensities and potentially shocks, executing this approach is problematic because of the strict requirements imposed on hydrophone bandwidth, robustness, and size. To overcome these issues, a method has been proposed that relies on acoustic holography and simulations of nonlinear propagation based on the 3D Westervelt model. This approach has been applied to several therapy transducers including a multi-element phased array. Uncertainties in the approach can be evaluated for both model boundary conditions determined from linear holography and the nonlinear focusing gain achieved at high power levels. Neglecting hydrophone calibration uncertainties, errors associated with the holography technique remain less than about 10% in practice. To assess the accuracy of nonlinear simulations, results were compared to independent measurements of focal waveforms using a fiber optic probe hydrophone (FOPH). When relative calibration uncertainties between the capsule hydrophone and FOPH are mitigated, simulations and FOPH measurements agree within about 15% for peak pressures at the focus. [Work supported by NIH grants EB016118, EB007643, T32 DK007779, DK43881, and NSBRI through NASA NCC 9-58.]

10:00–10:20 Break

10:20

4aBA7. Cavitation characteristics in High Intensity Focused Ultrasound lesions. Gail ter Haar and Ian Rivens (Phys., Inst. of Cancer Res., Phys. Dept., Royal Marsden Hospital, Sutton, Surrey SM2 5PT, United Kingdom, gail.terhaar@icr.ac.uk)

The acoustic emissions recorded during HIFU lesions fall into three broad categories: those associated with non-inertial cavitation, those associated with inertial cavitation, and those linked with tissue water boiling. These three mechanisms can be linked with different lesion shapes, and with characteristic histological appearance. By careful choice of acoustic driving parameters, these effects may be studied individually.

10:40

4aBA8. The role of tissue mechanical properties in histotripsy tissue fractionation. Eli Vlaisavljevich, Charles Cain, and Zhen Xu (Univ. of Michigan, 1111 Nielsen Ct. Apt. 1, Ann Arbor, MI 48105, evlaisav@umich.edu)

Histotripsy is a therapeutic ultrasound technique that controls cavitation to fractionate tissue using short, high-pressure ultrasound pulses. Histotripsy has been demonstrated to successfully fractionate many different tissues, though stiffer tissues such as cartilage or tendon (Young's moduli >1 MPa) are more resistant to histotripsy-induced damage than softer tissues such as liver (Young's moduli ~9 kPa). In this work, we investigate the effects of tissue mechanical properties on various aspects of the histotripsy process including the pressure threshold required to generate a cavitation cloud, the bubble dynamics, and the stress–strain applied to tissue structures. Ultrasound pulses of 1–2 acoustic cycles at varying frequencies (345 kHz, 500 kHz, 1.5 MHz, and 3 MHz) were applied to agarose tissue phantoms and ex vivo bovine tissues with varying mechanical properties. Results demonstrate that the intrinsic threshold to initiate a cavitation cloud is independent of tissue stiffness and frequency. The bubble expansion is suppressed in stiffer tissues, leading to a decrease in strain to surrounding tissue and an increase in damage resistance. Finally, we investigate strategies to optimize histotripsy therapy for the treatment of tissues with specific mechanical properties. Overall, this work improves our understanding of how tissue properties affect histotripsy and will guide parameter optimization for histotripsy tissue fractionation.

11:00

4aBA9. Technical advances for histotripsy: Strategic ultrasound pulsing methods for precise histotripsy lesion formation. Kuang-Wei Lin, Timothy L. Hall, Zhen Xu, and Charles A. Cain (Univ. of Michigan, 2200 Bonisteel Blvd., Gerstacker, Rm. 1107, Ann Arbor, MI 48109, kwlin@umich.edu)

Conventional histotripsy uses ultrasound pulses longer than three cycles wherein the bubble cloud formation relies on the pressure-release scattering of the positive shock fronts from sparsely distributed single cavitation bubbles, making the cavitation event unpredictable and sometimes chaotic. Recently, we have developed three new strategic histotripsy pulsing techniques to further increase the precision of cavitation cloud and lesion formation. (1) Microtripsy: When applying histotripsy pulses shorter than three cycles, the formation of a dense bubble cloud only depends on the applied peak negative pressure (P-) exceeding an intrinsic threshold of the medium. With a P- not significantly higher than this, very precise sub-vocal-volume lesions can be generated. (2) Dual-beam histotripsy: A sub-threshold high-frequency pulse (perhaps from an imaging transducer) is enabled by a sub-threshold low-frequency pump pulse to exceed the intrinsic threshold and produces very precise lesions. (3) Frequency compounding: a near monopolar pulse can be synthesized using a frequency-compounding transducer (an array transducer consisting of elements with various resonant frequencies). By adjusting time delays for individual frequency components and allowing their principal negative peaks to arrive at the focus concurrently, a near monopolar pulse with a dominant negative phase can be generated (no complicating high peak positive shock fronts).

11:20

4aBA10. Histotripsy: Urologic applications and translational progress. William W. Roberts (Urology, Univ. of Michigan, 3879 Taubman Ctr., 1500 East Medical Ctr. Dr., Ann Arbor, MI 48109-5330, willrobe@umich.edu), Charles A. Cain (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), J. B. Fowlkes (Radiology, Univ. of Michigan, Ann Arbor, MI), Zhen Xu, and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Histotripsy is an extracorporeal ablative technology based on initiation and control of acoustic cavitation within a target volume. This mechanical form of tissue homogenization differs from the ablative processes employed by conventional thermoablative modalities and exhibits a number of unique features (non-thermal, high precision, real-time monitoring/feedback, and tissue liquefaction), which are potentially advantageous characteristics for ablative applications in a variety of organs and disease processes. Histotripsy has been applied to the prostate in canine models for tissue debulking as a therapy for benign prostatic hyperplasia and for ablation of ACE-1 tumors, a canine prostate cancer model. Homogenization of normal renal tissue as well as implanted VX-2 renal tumors has been demonstrated with histotripsy. Initial studies assessing tumor metastases in this model did not reveal metastatic potentiation following mechanical homogenization by histotripsy. Enhanced understanding of cavitation and methods for acoustic control of the target volume are being refined in tank studies for treatment of urinary calculi. Development of novel acoustic pulsing strategies, refinement of technology, and enhanced understanding of cavitational bioeffects are driving pre-clinical translation of histotripsy for a number of applications. A human pilot trial is underway to assess the safety of histotripsy as a treatment for benign prostatic hyperplasia.

11:40

4aBA11. Boiling histotripsy of the kidney: Preliminary studies and predictors of treatment effectiveness. George R. Schade, Adam D. Maxwell (Dept. of Urology, Univ. of Washington, 5555 14th Ave. NW, Apt 342, Seattle, WA 98107, grschade@uw.edu), Tatiana Khokhlova (Dept. of Gastroenterology, Univ. of Washington, Seattle, WA), Yak-Nam Wang, Oleg Sapoznikov, Michael R. Bailey, and Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Boiling histotripsy (BH), an ultrasound technique to mechanically homogenize tissue, has been described in *ex vivo* liver and myocardium. As a noninvasive, non-thermal based approach, BH may have advantages over clinically available thermal ablative technologies for renal masses. We aimed to characterize BH exposures in human and porcine *ex vivo* kidneys using a 7-element 1 MHz transducer (duty factor 1–3%, 5–10 ms pulses, 98 MPa *in situ* shock amplitude, 17 MPa peak negative). Lesions were successfully created in both species, demonstrating focally homogenized tissue above treatment thresholds (pulse number) with stark transition between treated and untreated cells on histologic assessment. Human tissue generally required more pulses to produce similar effect compared to porcine. Similarly, kidneys displayed tissue specific resistance to BH with increasing resistance from cortex to medulla to the collecting system. Tissue properties that predict resistance to renal BH were evaluated demonstrating correlation between tissue collagen content and tissue resistance. Subsequently, the impact of intervening abdominal wall and ribs on lesion generation *ex vivo* was evaluated. “Transabdominal” and “transcostal” treatment required approximately 5- and 20-fold greater acoustic power, respectively, to elicit boiling vs. no intervening tissue. [Work supported by NIH T32DK007779, R01EB007643, K01EB015745 and NSBRI through NASA NCC 9-58.]

THURSDAY MORNING, 30 OCTOBER 2014 — MARRIOTT 9/10, 8:30 A.M. TO 11:15 A.M.

Session 4aEA

Engineering Acoustics: Acoustic Transduction: Theory and Practice I

Richard D. Costley, Chair
Geotechnical and Structures Lab., U.S. Army Engineer Research & Development Center, 3909 Halls Ferry Rd, Vicksburg, MS 39180

Contributed Papers

8:30

4aEA1. Historic transducers: Balanced armature receiver (BAR). Jont B. Allen (ECE, Univ. of Illinois, Urbana-Champaign, Urbana, IL) and Noori Kim (ECE, Univ. of Illinois, Urbana-Champaign, 1085 Baytowner df 11, Champaign, IL 61822, nkim13@illinois.edu)

The oldest telephone receiver is the Balanced Armature Receiver (BAR) type, and it is still in use. The original technology goes back to the invention of telephone receiver by A. G. Bell in 1876. Attraction and release of the armature are controlled by the current from the coils, which generates electromagnetic fields [Hunt (1954) Chapter 7, and Beranek and Mellow (2014)]. As the electrical current goes into the electric terminal of the receiver, it generates an AC magnetic field which direction is perpendicular to the current. Due to the polarity between the permanent (DC) magnet field and the generated AC magnetic field, an armature (which sits within the core of the coil and the magnet) feels a force. The very basic principles for explaining this movement in a gyrator, a fifth circuit element introduced by Tellegen in 1948, along with an inductor, a capacitor, a resistor, and a transformer. This component represents the anti-reciprocal characteristic of the system. This study is starting from comparing the BAR type receiver to the moving-coil loud speaker. We believe that this work will provide a fundamental and clear insight into this type of BAR system.

8:45

4aEA2. Radiation from wedges of a power law profile. Marcel C. Remillieux, Brian E. Anderson, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, mcr1@lanl.gov)

The large impedance contrast between bulk piezoelectric disks and air does not allow for efficient coupling of sound radiation from the piezoelectric into air. Here, we present the idea of using wedges of power law profiles to more efficiently radiate sound into air. Wedges of power law profiles have been used to provide absorption of vibrational energy in plates, but their efficient radiation of sound into air has not been demonstrated. We present numerical modeling and experimental results to demonstrate the concept. The wedge shape provides a gradual impedance contrast as the wave travels down the tapering of the wedge, while the wave speed also continually slows down. For an ideal wedge that tapers down to zero thickness, the waves become trapped at the tip and the vibrational energy can only radiate into the surrounding air. [This work was supported by institutional support [Laboratory Directed Research and Development (LDRD)] at Los Alamos National Laboratory.]

9:00

4aEA3. The self-sustained oscillator as an underwater low frequency projector: Progress report. Andrew A. Acquaviva and Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., c/o Steve Thompson, N-249 Millennium Sci. Complex, University Park, PA, acquavaa@gmail.com)

Wind musical instruments are examples of pressure operated self-sustained oscillators that act as acoustic projectors. Recent studies have shown that this type of self-sustained oscillator can also be implemented underwater as a low frequency projector. However, the results of the early feasibility studies were complicated by the existence of cavitation in the high pressure region of the resonator. A redesign has eliminated the cavitation and allows better comparison with analytical calculations.

9:15

4aEA4. Design and testing of an underwater acoustic Fresnel zone plate diffractive lens. David C. Calvo, Abel L. Thangawng, Michael Nicholas, and Christopher N. Layman, Jr. (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, david.calvo@nrl.navy.mil)

Fresnel zone plate (FZP) lenses offer a means of focusing sound based on diffraction in cases where the thickness of conventional lenses may be impractical. A binary-profile FZP for underwater use featuring a center acoustically opaque disk with alternating transparent and opaque annular regions was fabricated to operate nominally at 200 kHz. The overall diameter of the lens was 13 in. and consisted of 13 opaque annuli. The opaque regions were 3 mm thick and made from silicone rubber with a high concentration of gas voids. These regions were bonded to an acoustically transparent silicone rubber substrate film that was 1 mm thick. The FZP was situated in a frame and tested in a 5 x 4 x 4 cu. ft. ultrasonic tank using a piston source for insonification. The measured focal distance for normal incidence of 12.5 cm agreed with finite-element predictions taking into account the wavefront curvature of the incident field which had to be included given the finite dimensions of tank. The focal gain was measured to be 20 dB. The radius to the first null at the focal plane was approximately 4 mm, which agreed with theoretical resolution predictions. [Work sponsored by the Office of Naval Research.]

9:30

4aEA5. Acoustical transduction in two-dimensional piezoelectric array. Ola Nusierat, 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 acoustical transduction in an array of ferroelectric domains with alternating piezoelectric coefficients is characterized by multi-frequency resonances, which occur at the boundary of the acoustic Brillouin zone (ABZ). The resonances correspond to two successive domain excitations in the first and second ABZ correspondingly, where the speed of ultrasound is somewhat different. An important parameter for acoustical transduction is the electric impedance Z. The results of the theoretical and experimental investigations of Z in a periodically poled LiNbO3 are presented. The magnitude and phase of Z depend on the array parameters including domain resonance frequency and domain number; Z of arrays consisting of up to 88 0.45-mm-long domains in the zx-cut crystal are investigated. The strong changes in Z-magnitude and phase are observed in the range of 3–4 MHz. The two resonance zones are within 3.33 ± 0.05 MHz and 3.67± 0.05 MHz. The change in domain number influences Z and its phase. By varying the number of inversely poled domains and resonance frequencies, one can significantly control/change the electrical impedance of the multidomain array. The findings may be used for developing new acoustic sensors and transducers.

9:45

4aEA6. A non-conventional acoustic transduction method using fluidic laminar proportional amplifiers. Michael V. Scanlon (RDRL-SES-P, Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

Pressure sensing using fluidic laminar proportional amplifiers (LPAs) was developed at Harry Diamond Laboratories in the late 1970s and was applied to acoustic detection and amplification. LPAs use a partially constrained laminar jet of low-pressure air as the sensing medium, which is deflected by the incoming acoustic signal. LPA geometries enable pressure gain by focusing incoming pressure fluctuations at the jet's nozzle exit, thereby applying leverage to create jet deflection over its short transit toward a splitter. With no input signal, the jet is not deflected and downstream pressures on both sides of the splitter are equal. A differential input signal of magnitude one, referenced to ambient pressure balancing the opposite side of the jet, produces an differential output signal of magnitude ten. This amplified signal can be differentially fed into the inputs on both sides of the next LPA for additional gain. By cascading LPAs together, very small signals can be amplified a large amount. Originally, a DC pressure amplifier, LPAs have exceptional infrasound response, and excellent sensitivity since there is no mass or stiffness associated with a diaphragm, and is matched to the environment. Standard microphones at the output ports can take advantage of increased sensitivity and gain.

10:00–10:15 Break

10:15

4aEA7. Investigation of piezoelectric bimorph bender transducers to generate and receive shear waves. Andrew R. McNeese, Kevin M. Lee, Megan S. Ballard, Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), and R. Daniel Costley (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

This paper further demonstrates the ability of piezoceramic bimorph bender elements to preferentially generate and receive near-surface shear waves for in situ sediment characterization measurements, in terrestrial as well as marine clay soils. The bimorph elements are housed in probe transducers that can manually be inserted into the sediment and are based on the work of Shirley [J. Acoust. Soc. Am. 63(5), 1643–1645 (1978)] and of Richardson *et al.* [Geo.—Marine Letts. 196–203 (1997)]. The transducers can discretely generate and receive horizontally polarized shear waves, within their bimorph directivity patterns. The use of multiple probes allows one to measure the shear wave velocity and attenuation parameters in the sediment of interest. Measured shear wave data on a hard clay terrestrial soil, as well as on soft marine sediments, are presented. These parameters along with density and compressional wave velocity define the elastic moduli (Poisson's ratio, shear modulus, and bulk modulus) of the sediment, which are of interest in various areas of geophysics, underwater acoustics, and geotechnical engineering. Discussion will focus on use of the probes in both terrestrial and marine sediment environments. [Work supported by ARL:UT Austin.]

10:30

4aEA8. Multi-mode seismic source for underground application. abderrhamane ounadjela (sonic, Schlumberger, 2-2-1 Fuchinobe, Sagamihara, Sagamihara, Kanagawa 252-0206, Japan, ounadjela1@slb.com), Henri Pierre Valero, Jean christophe Auchere (sonic, Schlumberger, Sagamihara-Shi, Japan), and Olivier Moyal (sonic, Schlumberger, Clamart, France)

A new multi-mode downhole acoustic source has been designed to fulfill requirements of oil business. Three acoustic modes of radiation, i.e., monopole, dipole, and quadruple modes, respectively, are considered to assess the properties of the oil reservoir. Because of the geometry of the well it is challenging to design an efficient and effective powerful device. This new source uses an apparatus to convert the axial motion of the four motors distributed on the azimuth into a radial one. In order to make this conversion effective, the axial motion transformation into a radial one is performed thanks to a rod rolling on a cone; this conversion minimizes the loss by friction and is very effective. The conversion apparatus is also exploited to match the acoustic impedance of the surrounding medium. This new design is described in this paper as well as intensive modeling which allowed optimizing this multi-mode source device. Experimental data is in a good agreement with numerical modeling.

10:45

4aEA9. Sound characteristics of the caxirola when used by different uninstructed users. Talita Pozzer and Stephan Paul (UFSM, Tuiuti, 925. Apto 21, Santa Maria, RS 97015661, Brazil, talita.pozzer@eac.ufsm.br)

While originally developed to be the official musical instrument of the 2014 Soccer World Cup the caxirola was banned form the stadiums as could be thrown into the field by angry spectators. Nevertheless, outside the stadiums the caxirola was still used, thus an already started investigation into the acoustics of the caxirola was concluded. At a previous ASA meeting we presented the sound power level (SWL) of the caxirola only for the two most typical ways of use. Now we present data on the sound pressure level close to the user's ears (SPLcue) and the SWL, both measured in a reverberation room, from 30 subjects that used the caxirola according their understanding. It was found that the total SPLcue vary from 78 dB(A) up to 95 dB(A) and the global SWL of the caxirola varies from 72 dB until 84 dB. The distribution is not normal, them the SWL has 79 dB(A) as median, that is very similar of the result obtained at the previous study. The SPLcue and the SPL measured for calculating the SWL are differents. This probably due to the distance variation between the source and the ear of user causing a near field some times.

11:00

4aEA10. A micro-machined hydrophone using the piezoelectric-gate-of-field-effect-transistor for low frequency sounds. Min Sung, Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 416, POSTECH, San31, Hyoja-dong, Nam-gu, Pohang, Kyungbuk 790784, South Korea, smmath2@postech.ac.kr), Cheeyoung Joh (Underwater sensor Lab., Agency for Defense Development, Changwon, Kyungnam, South Korea,), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), Pohang, Kyungbuk, South Korea)

The micro-sized piezoelectric body for the miniaturized hydrophone is known to have the limits in low frequencies due to its high impedance and low sensitivity. In this study, a new transduction mechanism named as PiGoFET (piezoelectric gate of field effect transistor) is devised so that its application for the miniaturized hydrophone could overcome the limits of the micro-sized piezoelectric body. The PiGoFET transduction can be realized by combining a field effect transistor and a small piezoelectric body on its gate. A micro-machined silicon membrane of 2 mm diameter was connected to the small piezoelectric body so that acoustic pressure can apply appropriate forces on the body on the FET gate. The electric field from the deformed piezoelectric body modulates the channel current of FET directly, thus the transduction makes the sound pressure transferred to the source–drain current effectively at very low frequencies with micro-sized piezoelectric body. Under the described concept, a hydrophone was fabricated by micro-machining and calibrated using the comparison method in low frequencies to investigate its performance. [Research funded by MRCnD.]

4a THU. AM

Session 4aPAa

Physical Acoustics, Underwater Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, and Noise: Borehole Acoustic Logging and Micro-Seismics for Hydrocarbon Reservoir Characterization

Said Assous, Cochair
Geoscience, Weatherford, East Leake, Loughborough LE126JX, United Kingdom

David Eccles, Cochair
Weatherford, Geoscience, Loughborough, United Kingdom

Chair's Introduction—8:00

Invited Papers

8:05

4aPAa1. Generalized collar waves and their characteristics. Xiuming Wang, Xiao He, and Xiumei Zhang (State Key Lab. of Acoust., Inst. of Acoust. , 21 4th Northwestern Ring Rd., Hadian District, Beijing 100190, China, wangxm@mail.ioa.ac.cn)

A good acoustic logging while drilling (ALWD) tool is difficult to be designed because of collar waves that propagate along the tool. There always exist such acoustic waves in ALWD. The collar wave arrivals can strongly interfere with formation compressional waves in wave slowness picking up. In the past years, a considerable research work has been seen in suppressing collar waves in order to accurately pick up p- and s-wave slowness, and the obtained p- and s-wave slowness accuracy is still a problem. In this work, numerical and physical experiments are conducted to tackle collar wave propagation problems. And the collar wave propagation physics is elaborated and a generalized collar wave concept is proposed. It is shown that collar waves are much complex, and they consist of two kinds of collar waves, i.e., the direct collar waves and indirect collar waves. Both of these two collar waves make the ALWD data difficult to process for formation wave slowness picking up. Because of drilling string structures, the complicated collar waves cannot be effectively suppressed only with a groove isolator.

8:20

4aPAa2. Characterizing the nonlinear interaction of S (shear) and P (longitudinal) waves in reservoir rocks. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu), Thomas Gallot (Sci. Inst., Univ. of the Republic, Montevideo, Uruguay), Alison Malcolm, Stephen Brown, Dan Burns, and Michael Fehler (Earth Resources Lab., Massachusetts Inst. of Technol., Cambridge, MA)

The nonlinear elastic response of rocks is known to be caused by internal microstructure, particularly cracks and fluids. In order to quantify this nonlinearity, this paper presents a method for characterizing the interaction of two nonresonant traveling waves: a low-amplitude P-wave probe and a high-amplitude lower frequency S-wave pump with their particle motions aligned. We measure changes in the arrival time of the P-wave probe as it passes through the perturbation created by a traveling S-wave pump in a sample of room-dry Berea sandstone ($15 \times 15 \times 3$ cm). The velocity measurements are made at times too short for the shear wave to reflect back from the bottom of the sample and interfere with the measurement. The S-wave pump induces strains of $0.3—2.2 \times 10^{-6}$, and we observe changes in the P-wave probe arrival time of up to 100 ns, corresponding to a change in elastic properties of 0.2%. By changing the relative time delay between the probe and pump signals, we record the measured changes in travel time of the P-wave probe to recover the nonlinear parameters $\beta\sim -10^2$ and $\delta \sim\!-10^9$ at room-temperature. This work significantly broadens the applicability of dynamic acousto-elastic testing by utilizing both S and P traveling waves.

8:35

4aPAa3. A case study of multipole acoustic logging in heavy oil sand reservoirs. Peng Liu, Wenxiao Qiao, Xiaohua Che, Ruijia Wang, Xiaodong Ju, and Junqiang Lu (State Key Lab. of Petroleum Resources and Prospecting, China Univ. of Petroleum (Beijing), No. 18, Fuxue Rd., Changping District, Beijing, Beijing 102249, China, liupeng198712@126.com)

The multipole acoustic logging tool (MPAL) was tested in the heavy oil sand reservoirs of Canada. Compared with near shales, the P-wave slowness of heavy oil sands does not change obviously, with the value of about 125μs/ft; the dipole shear slowness decreases significantly to 275μs/ft. The heavy oil sands have a Vp/Vs value of less than 2.4. The slowness and amplitude of dipole shear wave are good lithology discriminators that have great differences between heavy oil sands and shales. The heavy oil sand reservoirs are anisotropic. The crossover phenomenon in the fast and slow dipole shear wave dispersion curves indicates that the anisotropy is induced by unbalanced horizontal stress in the region.

8:50

4aPAa4. Borehole sonic imaging applications. Jennifer Market (Weatherford, 19819 Hampton Wood Dr, Tomball, TX 77377, jennifer.market@weatherford.com)

The advent of azimuthal logging-while-drilling (LWD) sonic tools has opened up a surfeit of near-real time applications. Azimuthal images of compressional and shear velocities allow for geosteering, fracture identification, stress profiling, production enhancement, and 3D wellbore stability analysis. Combining borehole sonic images with electrical, gamma ray, and density images yields a detailed picture of the near- and far-wellbore nature of the stress field and resultant fracturing. A brief review of the physics of azimuthal sonic logging will be presented, paying particular attention to azimuthal resolution and depth of investigation. Examples of combined interpretations of sonic, density, and electrical images will be shown to illustrate fracture characterization, unconventional reservoir completion planning, and geosteering. Finally, recommendations for the optimized acquisition of borehole sonic images will be discussed.

Contributed Papers

9:05

4aPAa5. Numerical simulations of an electromagnetic actuator in a low-frequency range for dipole acoustic wave logging. Yinqiu Zhou, Penglai Xin, and Xiuming Wang (Inst. of Acoust., Chinese Acad. of Sci., 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, zhouyinqiu@mail.ioa.ac.cn)

In dipole acoustic logging, transducers are required to work in a low frequency range, such as 0.5–5 kHz, to measure shear wave velocities so as to accurately analyze the anisotropy parameters of formations. In this paper, an electromagnetic actuator is designed for more effective low-frequency excitations than conventional piezoelectric bender-bar transducers. A numerical model has been set up to simulate electromagnetic actuators to generate flexural waves. The Finite Element Method (FEM) has been applied to simulating the radiation modes and harmonic responses of the actuator in a fluid, such as air and water. In the frequency range of 0–5 kHz, the first ten vibration modes are simulated and analyzed. The simulation results of 3-D harmonic responses of the sound field, such as the deformation, acoustic sound pressure, and directivity pattern, have been conducted to evaluate the radiation performance. From the simulation results, it is concluded that the second asymmetric mode at 670 Hz could be excited more easily than the others. This oscillated-vibration mode is useful to be applied in a dipole source. The frequency response curve is broad and flat and the electromagnetic actuator is beneficial to generate the wideband signal in a required low frequency range, especially below 1 kHz.

9:20

4aPAa6. Phase moveout method for extracting flexural mode dispersion and borehole properties. Said Assous, David Eccles, and Peter Elkington (GeoSci., Weatherford, Weatherford, East Leake, Loughborough, United Kingdom, david.eccles@eu.weatherford.com)

Among the dispersive modes encountered in acoustic well logging applications is the flexural mode associated with dipole source excitations whose low frequency asymptote provides the only reliable means of determining shear velocity in slow rock formations. We have developed a phase moveout method for extracting flexural mode dispersion curves from with excellent velocity resolution; the method is entirely data-driven, but in combination with a forward model able to generate theoretical dispersion curves, we are able to address the inverse problem and extract formation and borehole properties in addition to the rock shear velocity. The concept is demonstrated using data from isotropic and anisotropic formations.

9:35

4aPAa7. Borehole acoustic array processing methods: A review. Said Assous and Peter Elkington (GeoSci., Weatherford, East Leake, Loughborough LE126JX, United Kingdom, said.assous@eu.weatherford.com)

In this talk, we review the different borehole acoustic array methods and compare their effectiveness with simulated and real waveform examples: Starting from the slowness time coherence (STC) method, weighted semblance method (WSS), and many other common dispersive processing approaches including: Prony's method, maximum entropy (ARMA) methods, and predictive array processing and Matrix pencil technique. We also discuss the Methods include phase minimization or coherency maximization and phase-based approaches.

9:50

4aPAa8. Classifying and removing monopole mode propagating through drill collar. Naoki Sakiyama (Schlumberger K.K., 2-18-3-406, Bessho, Hachio-ji 192-0363, Japan, NSakiyama@slb.com), Alain Dumont (Schlumberger K.K., Kawasaki, Japan), Wataru Izuhara (Schlumberger K.K., Inagi, Japan), Hiroaki Yamamoto (Schlumberger K.K., Kamakura, Japan), Makito Katayama (Schlumberger K.K., Yamato, Japan), and Takeshi Fukushima (Schlumberger K.K., Hachio-ji, Japan)

Understanding characteristics of the acoustic wave propagating through drill collars is important for formation evaluation with logging-while-drilling (LWD) sonic tools. Knowing the frequency-slowness information of different types of the wave propagating through the collar, we can minimize the unwanted wave propagating through the collar by processing and robustly identify formation compressional and shear arrivals. Extensional modes of the steel drill collar are generally dispersive and range from 180 μs/m to 400 μs/m depending on the frequency band. A fundamental torsional mode of the drill collar is nondispersive, but its slowness is sensitive to the geometry of the drill collar. Depending on the geometry and shear modulus of the material, the slowness of the torsional mode can be slower than 330 μs/m. For identifying slowness of the formation arrivals, those different slownesses of the wave propagating through the collar need to be identified separately from those of the wave propagating through formations. Examining various types of the acoustic wave propagating through a drill collar, we determined that the waves can be properly muted by processing for the semblance of waveforms acquired with LWD sonic tools.

10:05–10:20 Panel Discussion

Session 4aPAb

Physical Acoustics: Topics in Physical Acoustics I

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

William Slaton, Cochair
Physics & Astronomy, The University of Central Arkansas, 201 Donaghey Ave, Conway, AR 72034

Contributed Papers

10:30

4aPAb1. Faraday waves on a two-dimensional periodic substrate. C. T. Maki (Phys., Hampden-Sydney College, 1 College Rd., Hampden-Sydney, VA 23943, MakiC15@hsc.edu), Peter Rodriguez, Purity Dele-Oni, Pei-Chuan Fu, and R. Glynn Holt (Mech. Eng., Boston Univ., Boston, MA)

A vertically oscillating body of liquid will exhibit Faraday waves when forced above a threshold interface acceleration amplitude. The patterns and their wavelengths at driving frequencies of order 100 Hz are well known in the literature. However, wave interactions influenced by periodic structures on a driving substrate are less well-studied. We report results of a Faraday experiment with a specific periodically structured substrate in the strong coupling regime where the liquid depth is of the order of the structure height. We observe patterns and pattern wavelengths versus driving frequency over the range of 50–350 Hz. These observations may be of interest in situations where Faraday waves appear or are applied.

10:45

4aPAb2. Substrate interaction in ranged photoacoustic spectroscopy of layered samples. Logan S. Marcus, Ellen L. Holthoff, and Paul M. Pellegrino (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-SEE-E, Adelphi, MD 20783, loganmarcus@gmail.com)

Photoacoustic spectroscopy (PAS) is a useful monitoring technique that is well suited for ranged detection of condensed materials. Ranged PAS has been demonstrated using an interferometer as the sensor. Interferometric measurement of photoacoustic phenomena focuses on the measurement of changes in path length of a probe laser beam. That probe beam measures, without discrimination, the acoustic, thermal, and physical changes to the excited sample and the layer of gas adjacent to the surface of the solid sample. For layered samples, the photoacoustic response of the system is influenced by the physical properties of the substrate as well as the sample under investigation. We will discuss the affect that substrate absorption of the excitation source has on the spectra collected in PAS. We also discuss the role that the vibrational modes of the substrate have in photoacoustic signal generation.

11:00

4aPAb3. Difference frequency scattered waves from nonlinear interactions of a solid sphere. Chrisna Nguon (Univ. of Massachusetts Lowell, 63 Hemlock St., Dracut, MA 01826, chrisna_Nguon@student.uml.edu), Max Denis (Mayo Clinic, Rochester, MN), Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, Lowell, MA)

In this work, the generation of difference frequency waves arising from the interaction of dual-incident beams on a solid sphere is considered. The high-frequency incident beams induce a radiation force onto the fluid-saturated sphere causing the scatterer to vibrate. An analysis on the contribution between the difference frequency sound and radiation force pressure is of particular interest. The scattered pressure due to the two primary waves are obtained as solutions to the Kirchhoff–Helmholtz integral equation for the fluid–solid boundary. Due to the contrasting material properties between the host fluid and solid sphere, high-order approximations are used to evaluate the integral equation.

11:15

4aPAb4. Effect of surface irregularities on the stability of Stokes boundary. Katherine Aho, Jenny Au, Charles Thompson, and Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 University Ave, Lowell, MA 01854, katherine_aho@student.uml.edu)

In this work, we examine that impact that wall surface roughness plays on the stability of an oscillatory Stokes boundary layer. The temporal growth of three-dimensional disturbances excited by wall height variations is of particular interest. Floquet theory is used to identify linearly unstable region in parameter space. It is shown that disturbances become unstable at critical value of the Taylor number for a given surface curvature. The case of oscillatory flow in a two-dimensional rigid walled channel is considered in detail.

11:30

4aPAb5. Novel optoacoustic source for arbitrarily shaped acoustic wavefronts. Weiwei Chan, Yuanxiang Yang, Manish Arora, and Claus-Dieter Ohl (Phys. and Appl. Phys., Nanyang Technolog. Univ., Nanyang Link 21 School of Physical and Mathematical Sci. Nanyang Technolog. University, Singapore 637371, Singapore, chan0700@e.ntu.edu.sg)

We present a novel approach to generate arbitrary acoustic wavefronts using the optoacoustic effect on custom designed PDMS substrates. PDMS blocks are casted into the desired shape with a 3D-printed mold and coated with a layer of an optical absorber. Acoustic wavefront corresponding to the geometry of coated surface is generated by exposing this structure to nanosecond laser pulse (Nd:YAG, $\lambda = 532$ nm). For a spherical shell design, pressure pulses of amplitude up to 6.1 bar peak to peak and frequency >30 MHz could be generated. By utilizing other geometries, we focus the acoustic waves from different sections of the transmitter onto a single focal point at different time delay, thus permitting generation of double-peak acoustic pulse from a single laser pulse. Further modification of the structure permits designing of multi-foci, multi-peak acoustic pulses from a single optical pulse.

11:45

4aPAb6. Accuracy of local Kramers–Kronig relations between material damping and dynamic elastic properties. Tamas Pritz (Budapest Univ. of Technol. and Economics, Apostol u 23, Budapest 1025, Hungary, tampri@eik.bme.hu)

The local Kramers–Kronig (KK) relations are the differential form approximations of the general KK integral equations linking the damping properties (loss modulus or loss factor) and dynamic modulus of elasticity (shear, bulk, etc.) of linear solid viscoelastic materials. The local KK

relations are not exact; therefore, their accuracy is known to depend on the rate of frequency variations of material dynamic properties. The accuracy of the local KK relations is investigated in this paper under the assumption that the frequency dependence of the loss modulus obeys a simple power law. It is shown by analytic calculations that the accuracy of prediction of the local KK relations is better than 10% if the exponent in the loss modulus-frequency function is smaller than 0.35. This conclusion supports the result of an earlier numerical study. Some experimental data verifying the theoretical results will be presented. The conclusions drawn in the paper can easily be extended to acoustic wave propagation, namely to the accuracy of local KK relations between attenuation and dispersion of phase velocity.

THURSDAY MORNING, 30 OCTOBER 2014 MARRIOTT 1/2, 8:30 A.M. TO 12:00 NOON

Session 4aPP

Psychological and Physiological Acoustics: Physiological and Psychological Aspects of Central Auditory Processing Dysfunction I

Frederick J. Gallun, Cochair
National Center for Rehabilitative Auditory Research, Portland VA Medical Center, 3710 SW US Veterans Hospital Rd., Portland, OR 97239

Adrian KC Lee, Cochair
Box 357988, University of Washington, Seattle, WA 98195

Chair's Introduction—8:30

Invited Papers

8:35

4aPP1. Auditory processing disorder: Clinical and international perspective. David R. Moore (Commun. Sci. Res. Ctr., Cincinnati Children's Hospital, 240 Albert Sabin Way, Rm. S1.603, Cincinnati, OH 45229, david.moore2@cchmc.org)

APD may be considered a developmental, hearing or neurological disorder, depending on etiology, but in all cases, it is a listening difficulty without an abnormality of pure tone sensitivity. It has been variously defined as a disorder of the central auditory system associated with impaired spatial hearing, auditory discrimination, temporal processing, and performance with competing or degraded sounds. Clinical testing typically examines perception, intelligibility and ordering of both speech and non-speech sounds. While deficits in higher-order cognitive, communicative, and language functions are excluded in some definitions, recent consensus accepts that these functions may be inseparable from active listening. Some believe that APD in children is predominantly or exclusively cognitive in origin, while others insist that true APD has its origins within the auditory brainstem. However, children or their carers presenting at clinics typically complain of difficulty hearing speech in noise, remembering or understanding instructions, and attending to sounds. APD usually occurs alongside other developmental disorders (e.g., language impairment) and may be indistinguishable from them. Consequently, clinicians are uncertain how to diagnose or manage APD; both test procedures and interventions vary widely, even within a single clinic. Effective remediation primarily consists of improving the listening environment and providing communication devices.

9:05

4aPP2. Caught in the middle: The central problem in diagnosing auditory-processing disorders in adults. Larry E. Humes (Indiana Univ., Dept. Speech & Hearing Sci., Bloomington, IN 47405-7002, humes@indiana.edu)

It is challenging to establish the existence of higher-level auditory-processing disorders in military veterans with mild Traumatic Brain Injury (TBI). Yet, mild TBI appears to be a highly prevalent disorder among U.S. veterans returning from recent military conflicts in Iraq and Afghanistan. Recent prevalence estimates for mild TBI, for example, among these military veterans have suggest a rate of 7–9% [Carlson, K.F. *et al.* (2011), "Prevalence, assessment and treatment of mild Traumatic Brain Injury an Posttraumatic Stress Disorder: a systematic review of the evidence," J. Head Trauma Rehabil., 26, 103–115]. A key factor in diagnosing central components for auditory-processing disorders may lie in the potentially confounding influences of concomitant peripheral auditory and cognitive dysfunction in many veterans with TBI. This situation is strikingly similar to that observed in many older adults. Many older adults, for example, exhibit peripheral hearing loss and typical cognitive-processing deficits often associated with healthy aging. These concomitant problems make the diagnosis of centrally located auditory-processing problems in older adults extremely difficult. After building a case for many similarities between young veterans with mild TBI and older adults with presbycusis, this presentation will focus on several of the lessons learned from research with older adults. [Work supported, in part, by research grant R01 AG008293 from the NIA.]

9:35

4aPP3. Lack of a coherent theory limits the diagnosis and prognostic value of the central auditory processing disorder. Anthony T. Cacace (Commun. Sci. & Disord., Wayne State Univ., 207 Rackham, 60 Farnsworth, Detroit, MI 48202, cacacea@wayne.edu) and Dennis J. McFarland (Lab. of Neural Injury and Repair, Wadsworth Labs, NYS Health Dept., Albany, NY)

Spanning almost 6 decades, CAPD, defined as a modality specific perceptual dysfunction not due to peripheral hearing loss, still remains controversial and requires further development if it is to become a useful clinical entity. Early attempts to quantify the effects of central auditory nervous system lesions based on the use of filtered-speech material, dichotic presentation of digits, and various non-speech tests have generally been abandoned due to lack-of-success. Site-of-lesion approaches have given way to functional considerations whereby attempts to understand underlying processes, improve specificity-of-diagnosis, and delineate modality-specific (auditory) disorders from "non-specific supramodal dysfunctions" like those related to attention and memory have begun to fill the gap. Furthermore, because previous work was generally limited to auditory tasks alone, functional dissociations could not be established and consequently, the need to show the modality-specific nature of the observed deficits has been compromised; further limiting progress in this area. When viewed as a whole, including information from consensus conferences, organizational guidelines, representative studies, etc., what is conspicuously absent is a well-defined theory that permeates all areas of this domain, including the neural substrates of auditory processing. We will discuss the implications of this shortcoming and propose ways to move forward in a meaningful manner.

10:05–10:30 Break

10:30

4aPP4. Cochlear synaptopathy and neurodegeneration in noise and aging: Peripheral contributions to auditory dysfunction with normal thresholds. Sharon G. Kujawa (Dept. of Otology and Laryngology, Harvard Med. School and Massachusetts Eye and Ear Infirmary, Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114, sharon_kujawa@meei.harvard.edu)

Declining auditory performance in listeners with normal audiometric thresholds is often attributed to changes in central circuits, based on the widespread view that normal thresholds indicate a lack of cochlear involvement. Recent work in animal models of noise and aging, however, demonstrates that there can be functionally important loss of sensory inner hair cell—afferent fiber communications that go undetected by conventional threshold metrics. We have described a progressive cochlear synaptopathy that leads to proportional neural loss with age, well before loss of hair cells or age-related changes in threshold sensitivity. Similar synaptic and neural losses occur after noise, even when thresholds return to normal. Since the IHC-afferent fiber synapse is the primary conduit for information to flow from the cochlea to the brain, and since each of these cochlear nerve fibers makes synaptic contact with one inner hair cell only, these losses should have significant perceptual consequences, even if thresholds are preserved. The prevalence of such pathology in the human is likely to be high, underscoring the importance of considering peripheral status when studying central contributions to auditory performance declines. [Research supported by R01 DC 008577 and P30 DC 05029.]

11:00

4aPP5. Quantifying supra-threshold sensory deficits in listeners with normal hearing thresholds. Barbara Shinn-Cunningham (Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu), Hari Bharadwaj, Inyong Choi, Hannah Goldberg (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA), Salwa Masud, and Golbarg Mehraei (Speech and Hearing BioSci. and Technol., Harvard/MIT, Boston, MA)

There is growing suspicion that some listeners with normal-hearing thresholds may be suffering from a specific form of sensory deficit—a loss of afferent auditory nerve fibers. We believe such deficits manifest behaviorally in conditions where perception depends upon precise spectro-temporal coding of supra-threshold sound. In our lab, we find striking inter-subject differences in perceptual ability even among listeners with normal hearing thresholds who have no complaints of hearing difficulty and have never sought clinical intervention. Among such ordinary listeners, those who perform relatively poorly on selective attention tasks (requiring the listener to focus on one sound stream presented amidst competing sound streams) also exhibit relatively weak temporal coding in subcortical responses and have poor thresholds for detecting fine temporal cues in supra-threshold sound. Here, we review the evidence for supra-threshold hearing deficits and describe measures that reveal this sensory loss. Given our findings in ordinary adult listeners, it stands to reason that at least a portion of the listeners who are diagnosed with central auditory processing dysfunction may suffer from similar sensory deficits, explaining why they have trouble communicating in many everyday social settings.

11:30

4aPP6. Neural correlates of central auditory processing deficits in the auditory midbrain in an animal model of age-related hearing loss. Joseph P. Walton (Commun. Sci. and Disord., Univ. of South Florida, 4202 Fowler Ave., PCD 1017, Tampa, FL 33620, jwalton1@usf.edu)

Age-related hearing loss (ARHL), clinically referred to as presbycusis, affects over 10 million Americans and is considered to be the most common communication disorder in the elderly. Presbycusis can be associated with at least two underlying etiologies, a decline in cochlear function resulting in sensorineural hearing loss, and deficits in auditory processing within the central auditory system. Previous psychoacoustic studies have revealed that aged human listeners display deficits in temporal acuity that worsen with the addition of background noise. Spectral and temporal acuity is essential for following the rapid changes in frequency and intensity that comprise most natural sounds including speech. The perceptual analysis of complex sounds depends to a large extent on the ability of the auditory system to follow and even sharpen neural encoding of rapidly changing acoustic signals, and the inferior colliculus (IC) is a key auditory nucleus involved in temporal and spectral processing. In this talk, I will review neural correlates of temporal and signal-in-noise processing at the level of the auditory midbrain in an animal model of ARHL. Understanding the neural substrate of these perceptual deficits will assist in its diagnosis and rehabilitation, and be crucial to further advances in the design of hearing aids and therapeutic interventions.

Session 4aSCa

Speech Communication: Subglottal Resonances in Speech Production and Perception

Abeer Alwan, Cochair
Dept. of Electrical Eng., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095

Steven M. Lulich, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

4aSCa1. The role of subglottal acoustics in speech production and perception. Mitchell Sommers (Indiana Univ., Saint Louis, MS), Abeer Alwan (Psych., Washington Univ., Los Angeles, CA), and Steven Lulich (Psych., Washington Univ., Dept. of Speech and Hearing Sci., Indiana University, Bloomington, IN, slulich@indiana.edu)

In this talk, we present an overview of subglottal acoustics, with emphasis on the significant anatomical structures that define subglottal resonances, and we present results from our experiments incorporating subglottal resonances into automatic speaker normalization and speech recognition technologies. Speech samples used in the modeling and perception studies were obtained from a new speech corpus (the UCLA-WashU subglottal database) of simultaneous microphone and (subglottal) accelerometer recordings of 50 adult speakers of American English (AE). We will discuss new findings about the Young's Modulus of tracheal soft tissue, the viscosity of tracheal cartilage, and the effect of going from a circular cross-section to a rectangular cross-section in the conus elasticus. We also present results from studies demonstrating a small, but significant, role of subglottal resonances in discriminating speaker height and of the interaction between subglottal resonances and formants in height discrimination.

8:25

4aSCa2. The effect of subglottal acoustics on vocal fold vibration. Ingo R. Titze (National Ctr. for Voice and Speech, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@ncvs2.org) and Ingo R. Titze (Dept. of Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA)

Acoustic pressures above and below the vocal folds produce a push-pull action on the vocal folds which can either help or hinder vocal fold vibration. The key variable is acoustic reactance, the energy-storage part of the complex acoustic impedance. For the subglottal airway, inertive (positive) reactance does not help vocal fold vibration, but helps to skew the glottal airflow waveform for high frequency harmonic excitation. Compliant (negative) reactance, on the contrary, helps vocal fold vibration but does not skew the waveform. Thus, the benefit of subglottal reactance is mixed. For supraglottal reactance, the benefit is additive. Inertive supraglottal reactance helps vocal fold vibration and skews the waveform, whereas compliant supraglottal reactance does neither. The effects will be demonstrated with source-filter interactive simulation.

8:45

4aSCa3. Impact of subglottal resonances on bifurcations and register changes in laboratory models of phonation. David Berry, Juergen Neubauer, and Zhaoyan Zhang (Surgery, UCLA, 31-24 Rehab, Los Angeles, CA 90095-1794, daberry@ucla.edu)

Many laboratory studies of phonation have failed to fully specify the subglottal system employed during research. Many of these same studies have reported a variety of nonlinear phenomena, such as bifurcations and vocal register changes. While such phenomena are often presumed to result from changes in the biomechanical properties of the larynx, such phenomena may also be a manifestation of coupling between the voice source and the subglottal tract. Using laboratory models of phonation, a variety of examples will be given of nonlinear phenomena induced by both laryngeal and subglottal mechanisms. Moreover, using tracheal tube lengths commonly reported in the literature, it will be shown that most of the nonlinear phenomena commonly reported in voice production may be replicated solely on the acoustical resonances of the subglottal system. Finally, recommendations will be given regarding the experimental design of laboratory experiments which may allow laryngeally induced bifurcations to be distinguished from subglottally induced bifurcations.

9:05–9:25 Break

9:25

4aSCa4. Subglottal ambulatory monitoring of vocal function to improve voice disorder assessment. Robert E. Hillman, Daryush Mehta, Jarrad H. Van Stan (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, One Bowdoin Square, 11th Fl., Boston, MA 02114, daryush.mehta@alum.mit.edu), Matias Zanartu (Dept. of Electron. Eng., Universidad Tecnica Federico Santa Mariıa, Valparaiso, Chile), Marzyeh Ghassemi, and John V. Guttag (Comput. Sci. and Artificial Intelligence Lab., Massachusetts Inst. of Technol., Cambridge, MA)

Many common voice disorders are chronic or recurring conditions that are likely to result from inefficient and/or abusive patterns of vocal behavior, referred to as vocal hyperfunction. The clinical management of hyperfunctional disorders would be greatly enhanced by the ability to monitor and quantify detrimental vocal behaviors during an individual's activities of daily life. This presentation will provide an update about ongoing work that is using a miniature accelerometer on the subglottal neck surface to collect a large set of ambulatory data on patients with hyperfunctional voice disorders (before and after treatment) and matched control subjects. Three types of analysis approaches are being employed in an effort to identify the best set of measures for differentiating among hyperfunctional and normal patterns of vocal behavior: (1) previously developed ambulatory measures of vocal function that include vocal dosages; (2) measures based on estimates of glottal airflow that are extracted from the accelerometer signal using a vocal system model, and (3) classification based on machine learning approaches that have been used successfully in analyzing long-term recordings of other physiologic signals (e.g., electrocardiograms).

9:45

4aSCa5. Do subglottal resonances lead to quantal effects resulting in the features [back] and [low]?: A review. Helen Hanson (ECE Dept., Union College, 807 Union St., Schenectady, NY 12308, helen.hanson@alum.mit.edu) and Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA)

A question of general interest is why languages have the sound categories that they do. K. N. Stevens proposed the Quantal Theory of phonological contrasts, suggesting that regions of discontinuity in the articulatory-acoustic mapping serve as category boundaries. H. M. Hanson and K. N. Stevens [Proc. ICPhS, 182–185, 1995] modeled the interaction of subglottal resonances with the vocal-tract filter, showing that when a changing supraglottal formant strays into the territory of a stationary tracheal formant, a discontinuity in supraglottal formant frequency and attenuation of the formant peak occurs. They suggested that vowel space and quality could thus be affected. K. N. Stevens [*Acoustic Phonetics*, MIT Press, 1998] went further, musing that because the first and second subglottal resonances lead to instabilities in supraglottal formant frequency and amplitude, vowel systems would benefit by avoiding vowels with formants at these frequencies. Avoiding the first subglottal resonance would naturally lead to the division of vowels into those with a low vs. non-low tongue body; avoiding the second would lead to the division of vowels into those having a back vs. front tongue body. We will review subsequent research that offers substantial support for this hypothesis, justifying inclusion of the effects of subglottal resonances in phonological models.

Contributed Paper

10:05

4aSCa6. Relationship between lung volumes and subglottal resonances. Natalie E. Duvanenko (Speech and Hearing Sci., Indiana Univ., 2416 Cibuta Court, West Lafayette, IN 47906, nduvanen@umail.iu.edu) and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Subglottal resonances are dependent the anatomical structure of the lungs, but efforts to detect changes in subglottal resonances throughout an utterance have failed to show any effect of lung volume. In this study, we present the results of an experiment investigating the relationship between lung volumes and subglottal resonances. The pulmonary subdivisions for several speakers were established using a whole-body plethysmograph. Subsequently, lung volume and subglottal resonances were recorded simultaneously using a spirometer and an accelerometer while the speakers produced long sustained vowels.

Session 4aSCb

Speech Communication: Learning and Acquisition of Speech (Poster Session)

Maria V. Kondaurova, Chair
Otolaryngology – Head & Neck Surgery, Indiana University School of Medicine, 699 Riley Hospital Drive – RR044, Indianapolis, IN 46202

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

8:00

4aSCb1. Labels facilitate the learning of competing abstract perceptual mappings. Shannon L. Heald, Nina Bartram, Brendan Colson, and Howard C. Nusbaum (Psych., Univ. of Chicago, 5848 S. University Ave., B406, Chicago, IL 60637, smbowdre@uchicago.edu)

Listeners are able to quickly adapt to synthetic speech, even though it contains misleading and degraded acoustic information. Previous research has shown that testing and training on a given synthesizer using only novel words leads listeners to form abstract or generalized knowledge for how that particular synthesizer maps different acoustic patterns onto their pre-existing phonological categories. Prior to consolidation, this knowledge has been shown to be susceptible to interference. Given that labels have been argued to stabilize abstract ideas in working memory and to help learners form category representations that are robust against interference, we examined how learning for a given synthesizer is affected by labeled or unlabeled immediate training on an additional synthesizer, which uses a different acoustic to phonetic mapping. We demonstrated that the learning of an additional synthesizer interferes with the retention of a previously learned synthesizer but that this is ameliorated if the additional synthesizer is labeled. Our findings indicate that labeling may be important in facilitating daytime learning for competing abstract perceptual mappings prior to consolidation and suggests that speech perception may be best understood through the lens of perceptual categorization.

4aSCb2. When more is not better: Variable input in the formation of robust word representations. Andrea K. Davis (Linguist, Univ. of Arizona, 1076 Palomino Rd., Cloverdale, CA 95425, davisak@email.arizona.edu) and LouAnn Gerken (Linguist, Univ. of Arizona, Tucson, AZ)

A number of studies with infants and with young children suggest that hearing words produced by multiple talkers helps learners to develop more robust word representations (Richtsmeier *et al*., 2009; Rost & McMurray, 2009, 2010). Native adult learners, however, do not seem to derive the same benefit from multiple talkers. A word-learning study with native adults was conducted, and a second study with second language learners will have been completed by this fall. Native-speaking participants learned four new minimal English-like minimal pair words either from a single talker or from multiple talkers. They were then tested with (a) a perceptual task, in which they saw the two pictures corresponding to a minimal pair, heard one of the pair, and had to choose the picture corresponding to the word they heard; (b) a speeded production task, in which they had to repeat the words they had just learned as quickly as possible. Unlike infants, the two groups did not differ significantly in perceptual accuracy. However, the single talker group had significantly higher variance in the speeded production task. It is hypothesized that this greater variance is due to individual differences in learning strategies, which are masked when learning from multiple talkers.

4aSCb3. A comparison of acoustic and perceptual changes in children's productions of American English /r/. Sarah Hamilton (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Casey Keck (Commun. Sci. and Disord., Univ. of Cincinnati, 408 Glengarry Way, Fort Wright, KY 41011, stewarce@mail.uc.edu), and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Speech-language pathologists rely primarily on their perceptual judgments when evaluating whether children have made progress in speech sound therapy. Speech sound perception in normal listeners has been characterized as largely categorical, such that slight articulatory changes may go unnoticed unless they reach a specific acoustic signature assigned to a different category. While perception may be categorical, acoustic phenomena are largely measured in continuous units, meaning that there is a potential mismatch between the two methods of recording change. Clinicians, using perceptual categorization, commonly report that some children make no progress in therapy, yet acoustically, the children's productions may be shifting toward acceptable acoustic characteristics. Using subtle changes in the acoustic signal during therapy could potentially prevent these clients from being discharged due to a perceived lack of progress. This poster evaluates acoustic changes compared to perceptual changes in children's productions of the American English phoneme /r/ after receiving speech therapy using ultrasound supplemented with telepractice home practice. Preliminary data indicate that there are significant differences between participants' acoustic values of /r/ and perceptual ratings by clinicians.

4aSCb4. Perceptual categorization of /r/ for children with residual sound errors. Sarah M. Hamilton, Suzanne Boyce, and Lindsay Mullins (Commun. Sci. and Disord., Univ. of Cincinnati, 3433 Clifton Ave., Cincinnati, OH 45220, hamilsm@mail.uc.edu)

Many studies have found that children with resistant speech sound errors (RSSD) show (1) atypical category boundaries, and (2) difficulty identifying whether their own productions are correct or misarticulated. Historically, perceptual category discrimination tests use synthesized speech representing incremental change along an acoustic continuum, while tests of a child's self-perception are confined to categorical correct vs. error choices. Thus, it has not been possible to explore the boundaries of RSSD children's categorical self-perception in any detail or to customize perceptual training for therapeutic purposes. Following an observation of Hagiwara (1995), who noted that typical speakers show F3 values for /r/ between 80% and 60% of their average vowel F3, Hamilton et al (2014) found that this threshold largely replicates adult listener judgments, such that productions above and below the 80% threshold sounded consistently "incorrect" or "correct," but that productions closest to the 80% threshold were given more ambiguous judgments. In this study, we apply this notion of an F3 threshold to investigate whether children with RSD respond like adult listeners when presented with natural-speech stimuli along a continuum of correct and incorrect /r/. Preliminary results indicate that children with RSD do not make adult-like decisions when categorizing /r/ productions.

4aSCb5. A child-specific compensatory mechanism in the acquisition of English /s/. Hye-Young Bang, Meghan Clayards, and Heather Goad (Linguist, McGill Univ., 1085 Dr. Penfield, Montreal, QC H3A 1A7, Canada, hye-young.bang@mail.mcgill.ca)

This study examines corpus data involving word-initial [sV] productions from 79 children aged 2–5 (Edwards & Beckman 2008) in comparison with a corpus of word-initial [sV] syllables produced by 13 adults. We quantified target-like /s/ production using spectral moment analysis on the frication portion (high center of gravity, low SD, and low skewness). In adults, we found that higher vowels (low F1 after normalization) were associated with more target-like /s/ productions, likely reflecting a tighter constriction. In children, older subjects produced more target-like outputs overall. However, unlike adults, children's outputs before low vowels were more target-like, regardless of age. This is unexpected given the articulatory challenges of producing /s/ in low vowel contexts. Further investigation found that high F1 (low vowels) was associated with louder /s/ (relative to V) and more encroachment of sibilant noise on the following vowel (high harmonics-to-noise ratio). This finding suggests that young children may be increasing airflow during /s/ production to compensate for a less tight constriction when the jaw must lower for the following vowel. Thus, children may adopt a more accessible mechanism, different from adults, to compensate for their immature lingual gestures, possibly in an attempt to maximize phonological contrasts in word-initial position.

4aSCb6. Moving targets and unsteady states: "Shifting" productions of sibilant fricatives by young children. Patrick Reidy (Dept. of Linguist, The Ohio State Univ., 24A Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, patrick.francis.reidy@gmail.com)

The English voiceless sibilant /s/–/ʃ/ contrast is one that many children do not acquire until their adolescent years. This protracted acquisition may be due to the high level of articulatory control that is necessary to the successful production of an adult-like sibilant, which involves the coordination of lingual, mandibular, and pulmonic gestures. Poor coordination among these gestures can result in the acoustic properties of the noise source or the vocal tract filter changing throughout the timecourse of the frication, to the extent that the phonetic percept of the frication noise changes across its duration. The present study examined such "shifting" productions of sibilant fricatives by native English-acquiring two- through five-year-old children, which were identified from the Paidologos corpus as those productions where the interval of frication was transcribed phonetically as a sequence of fricative sounds. There were two types of shift in frication quality: (1) a gradual change in the resonant frequencies in the spectrogram, suggesting a repositioning of the oral constriction; and (2) an abrupt change in the level of the frication, suggesting a switch in the noise source. Work is underway to develop measures that differentiate these two types of shift, and that suggest their underlying articulatory causes.

4aSCb7. Effects of spectral smearing on sentence recognition by adults and children. Joanna H. Lowenstein (Otolaryngology-Head & Neck Surgery, Ohio State Univ., 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, lowenstein.6@osu.edu), Eric Tarr (Audio Eng. Technol., Belmont Univ., Nashville, TN), and Susan Nittrouer (Otolaryngology-Head & Neck Surgery, Ohio State Univ., Columbus, OH)

Children's speech perception depends on dynamic formant patterns more than that of adults. Spectral smearing of formants, as found with the broadened auditory filters associated with hearing loss, should disproportionately affect children because of this greater dependence on formant patterns. Making formants more prominent, on the other hand, may result in improved recognition. Adults (40) and children age 5 and 7 (20 of each age) listened to 75 four-word syntactically correct, semantically anomalous sentences processed so that excursions around the mean spectral slope were sharpened by 50% (making individual formants more prominent), flattened by 50% (smearing individual formants), or left unchanged. These sentences were presented to children and to half of the adults in speech-shaped noise at 0 dB SNR. The rest of the adults listened to the sentences at -3 dB SNR. Results indicate that all listeners did more poorly with the smeared formants, with 5-year-olds showing the largest decrement in performance at 0 dB SNR. However, adults at -3 dB SNR showed an even greater decrement in performance. Making formants more prominent did not improve recognition, perhaps due to harmonic-formant mismatches. Thus, there is reason to explore processing strategies that might enhance formant prominence for listeners with hearing loss.

4aSCb8. Acoustic-phonetic characteristics of older children's spontaneous speech in interactions in conversational and clear speaking styles. Valerie Hazan, Michèle Pettinato, Outi Tuomainen, and Sonia Granlund (Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

This study investigated (a) the acoustic-phonetic characteristics of spontaneous speech produced by talkers aged 9–14 years in an interactive (diapix) task with an interlocutor of the same age and gender (NB condition) and (b) the adaptations these talkers made to clarify their speech when speech intelligibility was artificially degraded for their interlocutor (VOC condition). Recordings were made for 96 child talkers (50 F, 46 M); the adult reference values came from the LUCID corpus recorded under the same conditions [Baker and Hazan, J. Acoustic. Soc. Am. 130, 2139–2152 (2011)]. Articulation rate, pause frequency, fundamental frequency, vowel area, and mean intensity (1–3 kHz range) were analyzed to establish whether they had reached adult-like values and whether young talkers showed similar clear speech strategies as adults in difficult communicative situations. In the NB condition, children (including the 13–14 year group) differed from adults in terms of their articulation rate, vowel area, median F0, and intensity. Child talkers made adaptations to their speech in the VOC condition, but adults and children differed in their use of F0 range, vowel hyperarticulation, and pause frequency as clear speech strategies. This suggests that further developments in speech production take place during later adolescence. [Work supported by ESRC.]

4aSCb9. Acoustic characteristics of infant-directed speech to normal-hearing and hearing-impaired twins with hearing aids and cochlear implants: A case study. Maria V. Kondaurova, Tonya R. Bergeson-Dana (Otolaryngol. – Head & Neck Surgery, Indiana Univ. School of Medicine, 699 Riley Hospital Dr. – RR044, Indianapolis, IN 46202, mkondaur@iupui.edu), and Neil A. Wright (The Richard and Roxelyn Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

The study examined acoustic characteristics of maternal speech to normal-hearing (NH) and hearing-impaired (HI) twins who received hearing aids (HAs) or a unilateral cochlear implant (CI). A mother of female-male NH twins (NH-NH; age 15.8 months), a mother of two male twins, one NH and another HI with HAs (NH-HA; age 11.8 months) and a mother of a NH female twin and a HI male twin with a CI (NH-CI; age 14.8 months) were recorded playing with their infants during three sessions across a 12-month period. We measured pitch characteristics (normalized F0 mean, F0 range, and F0 SD), utterance and pause duration, syllable number, and speaking rate. ANOVAs demonstrated that speech to NH-NH twins was characterized by lower, more variable pitch with greater pitch range as compared to speech to NH-HA and NH-CI pairs. Mothers produced more syllables, had faster speaking rate and longer utterance duration in speech to NH-NH than the other pairs. The results suggest that the pediatric hearing loss in one sibling affects maternal speech properties to both NH and HI infants in the same pair. Future research will investigate vowel space and lexical properties of IDS to three twin pairs as well as their language outcome measures.

4aSCb10. Effects of vowel position and place of articulation on voice onset time in children: Longitudinal data. Elaine R. Hitchcock (Dept. of Commun. Sci. and Disord., Montclair State Univ., 1515 BRd. St., Bloomfield, NJ 07444, hitchcocke@mail.montclair.edu) and Laura L. Koenig (Dept. of Commun. Sci. and Disord., Long Island Univ., Queens, NY)

Voice onset time (VOT) has been found to vary according to phonetic context, but past studies report varying magnitudes of effect, and no past work has evaluated the degree to which such effects are consistent over time for a single speaker. This study explores the relationships between vowel position, consonant place of articulation [POA], and voice onset time (VOT) in children, comparing the results to past adult work. VOT in CV/CVC words was measured in nine children ages 5;3–7;6 every two-four

weeks for 10 months, for a total of 18 sessions yielding approximately 18,000 tokens for analysis. Bilabial and velar cognate pairs targeted a front-back vowel difference (/i/-/u/, /e/-/o/), while alveolar cognate pairs targeted a mid high-low vowel difference (/o/-/ɑ/). VOT variability over time was also evaluated. Preliminary results suggest that POA yields a robust pattern of bilabial < alveolar < velar, but vowel effects are less clear. Vowel height shows the most obvious effect with consistently longer VOT values observed for mid high vowels. Front-back vowel comparisons yielded no obvious differences. On the whole, contextual variations based on POA and vowel context do not show clear correlations with overall VOT variation.

4aSCb11. Longitudinal data on the production of content versus function words in children's spontaneous speech. Jeffrey Kallay and Melissa A. Redford (Linguist, Univ. of Oregon, 1455 Moss St., Apt. 215, Eugene, Ohio 97403, jkallay@uoregon.edu)

Allen and Hawkins (1978; 1980) were among the first to note rhythmic differences in the speech of children and adults. Sirsa and Redford (2011) found that rhythmic differences between younger and older children's speech was best accounted for by age-related differences in function word production. In other on-going work (Redford, Kallay & Dilley) we found an effect of age on the perceived prominence of function words in children's speech, but no effect on content words. The current longitudinal study investigated the effect of word class (content versus function words) on the development of reduction in terms of syllable duration and pitch range (a correlate of accenting). Spontaneous speech was elicited for 3 years from 36 children aged 5; 2–6; 11 at time of first recording. There were effects of word class (content > function) and of time on median duration, but no interaction between these factors. The median duration decreased 13% in function words from the 1st to 3rd year; a similar decrease (15%) was found for content words. Pitch range only varied systematically with word class. Other spectral measures are being collected to further investigate the development of reduction in children's speech. [Work supported by NICHD.]

4aSCb12. Audiovisual speech integration development at varying levels of perceptual processing. Kaylah Lalonde (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, klalonde@indiana.edu) and Rachael Frush Holt (Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

There are multiple mechanisms of audiovisual (AV) speech integration with independent maturational time courses. This study investigated development of both basic perceptual and speech-specific mechanisms of AV speech integration by examining AV speech integration development across three levels of perceptual processing. Twenty-two adults and 24 6- to 8-year-old children completed three auditory-only and AV yes/no tasks varying only in the level of perceptual processing required to complete them: detection, discrimination, and recognition. Both groups demonstrated benefits from matched AV speech and interference from mismatched AV speech relative to auditory-only conditions. Adults, but not children, demonstrated greater integration effects at higher levels of perceptual processing (i.e., recognition). Adults seem to rely on both general perceptual mechanisms of speech integration that apply to all levels of perceptual processing and speech-specific mechanisms of integration that apply when making phonetic decisions and/or accessing the lexicon; 6- to 8-year-old children seem to rely only on general perceptual mechanisms of AV speech integration. The general perceptual mechanism allows children to attain the same degree of AV benefit to detection and discrimination as adults, but the lack of a speech-specific mechanism in children might explain why they attain less AV recognition benefit than adults.

4aSCb13. Developmental and linguistic factors of audiovisual speech perception across different masker types. Rachel Reetzke, Boji Lam, Zilong Xie, Li Sheng, and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas at Austin, The University of Texas at Austin, 2504A Whitis Ave., Austin, TX 78751, rreetzke@gmail.com)

Developmental and linguistic factors have been found to influence listeners' ability to recognize speech-in-noise. However, there is paucity of evidence exploring how these factors modulate speech perception in everyday listening situations, such as multisensory environments and backgrounds with informational maskers. This study assessed sentence recognition for 30 children (14 monolingual, 16 simultaneous bilingual; ages 6–10) and 31 adults (21 monolingual, ten simultaneous bilingual; ages 18–22). Our experimental design included three within-subject variables: (a) masker type: pink noise or two-talker babble, (b) modality: audio-only and audiovisual, and (c) signal-to-noise ratio (SNR): 0 to -16 dB. Results revealed that across both modalities and noise types, adults performed better than children, and simultaneous bilinguals performed similarly to monolinguals. The age effect was largest at the lowest SNRs of -12 and -16 dB in the audiovisual two-talker babble condition. These findings suggest that children experience greater difficulty in segregation of target speech in informational maskers relative to adults, even with audiovisual cues. This may provide evidence for children's less developed higher-level cognitive strategies in dealing with speech-in-noise (e.g., selective attention). Findings from the second analysis suggest that despite two competing lexicons, simultaneous bilinguals do not experience a speech perception-in-noise deficit relative to monolinguals.

4aSCb14. Experience-independent effects of matching and non-matching visual information on speech perception. D. Kyle Danielson, Alison J. Greuel, Padmapriya Kandhadai, and Janet F. Werker (Psych., Univ. of Br. Columbia, 2136 West Mall, Vancouver, BC V6T 1Z4, Canada, kdanielson@psych.ubc.ca)

Infants are sensitive to the correspondence between visual and auditory speech. Infants exhibit the McGurk effect, and matching audiovisual information may facilitate discrimination of similar consonant sounds in an infant's native language (e.g., Teinonen *et al*., 2008). However, because most existing research in audiovisual speech perception has been conducted using native speech sounds with infants in their first year of life, little work has explored whether this link between the auditory and visual modalities of speech perception arises due to experience with the native language. In the present set of studies, English-learning six- and ten-month-old infants are tested for discrimination of a non-English speech contrast following familiarization with matching and mismatching audiovisual speech. Furthermore, the looking fixation behaviors of the two age groups are compared between the two conditions. Although it has been demonstrated that infants in the younger age range attend preferentially to the eye region when viewing matched audiovisual speech and that infants in the older age range temporarily attend to the mouth region (Lewkowicz & Hansen-Tift, 2012), here deviations in this behavior for matching and mismatching non-native speech are examined (a link that has only been previously explored in the native language (Tomalski *et al*., 2013)).

4aSCb15. Switched-dominance bilingual speech production: Continuous usage versus early exposure. Michael Blasingame and Ann R. Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, mblasingame@u.northwestern.edu)

Switched dominance bilinguals (i.e., "heritage speakers," HS, with L2 rather than L1 dominance) have exhibited native-like heritage language (L1) sound perception (e.g., Korean three-way VOT contrast discrimination by Korean HS; Oh, Jun, Knightly, & Au, 2003) and sound production (e.g., Spanish VOT productions by Spanish HS; Au, Knightly, Jun, & Oh, 2002), but far from native-like proficiency in other aspects of L1 function, including morphosyntax (Montrul, 2010). We investigated whether native-like L1 sound production proficiency extended to heritage language sentence-in-noise intelligibility. We recorded English and Spanish sentences by Spanish HS (SHS) and monolingual English controls (English only). Native listeners of each language transcribed these recordings under easy (-4 dB SNR) and hard (-8 dB SNR) conditions. In easy conditions, SHS English and Spanish intelligibility were not significantly different, yet in hard conditions, SHS English intelligibility was significantly higher than SHS Spanish intelligibility. Furthermore, we observed no differences between SHS English and English-control intelligibility in both conditions. These results suggest for SHS, while early Spanish exposure provided some resistance to heritage language/L1 intelligibility degradation, the absence of continuous Spanish usage impacted intelligibility in severely degraded conditions. In contrast, the absence of early English exposure was entirely overcome by later English dominance.

4aSCb16. Genetic variation in catechol-O-methyl transferase activity impacts speech category learning. Han-Gyol Yi (Commun. Sci. and Disord., The Univ. of Texas at Austin, 2504 Whitis Ave., A1100, Austin, TX 78712, gyol@utexas.edu), W. T. Maddox (Psych., The Univ. of Texas at Austin, Austin, TX), Valerie S. Knopik (Behavioral Genetics, Rhode Island Hospital, Providence, RI), John E. McGeary (Providence Veterans Affairs Medical Ctr. , Providence, RI), and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

Learning non-native speech categories is a challenging task. Little is known about the neurobiology underlying speech category learning. In vision, two dopaminergic neurobiological learning systems have been identified: a rule-based reflective learning system mediated by the prefrontal cortex, wherein processing is under deliberative control, and an implicit reflexive learning system mediated by the striatum. During speech learning, successful learners initially use simple reflective rules but eventually transition to a multidimensional reflexive strategy during later learning. We use a neurocognitive-genetic approach to identify intermediate phenotypes that modulate reflective brain function and examine their effects on speech learning. We focus on the COMT Val158Met polymorphism, which is linked to altered prefrontal function. The COMT-Val variant catabolizes dopamine more rapidly and is linked to poorer performance on prefrontally-mediated tasks. Adults (Met-Met: = 40; Met-Val= 75; Val-Val = 54) learned to categorize non-native Mandarin tones over five blocks of feedback-based training. Learning rates were the highest for the Met-Met genotype; the Val-Val genotype was associated with poorer overall learning. Poorer learning indicates increased perseveration of reflective unidimensional rule use, thereby preventing the transition to the reflexive system. We conclude that genetic variation is an important source of individual differences in complex phenotypes such as speech learning.

THURSDAY MORNING, 30 OCTOBER 2014 — INDIANA G, 9:00 A.M. TO 10:00 A.M.

Session 4aSPa

Signal Processing in Acoustics: Imaging and Classification

Grace A. Clark, Chair

Grace Clark Signal Sciences, 532 Alden Lane, Livermore, CA 94550

Contributed Papers

9:00

4aSPa1. Optimal smoothing splines improve efficiency of entropy imaging for detection of therapeutic benefit in muscular dystrophy. Michael Hughes (Int. Med./Cardiology, Washington Univ. School of Medicine, 1632 Ridge Bend Dr., St Louis, MO 63108, mshatctrain@gmail.com), John McCarthy (Dept. of Mathematics, Washington Univ., St. Louis, MO), Jon Marsh (Int. Med./Cardiology, Washington Univ. School of Medicine, Saint Louis, MO), and Samuel Wickline (Dept. of Mathematics, Washington Univ., Saint Louis, MO)

We have reported previously on sensitivity comparisons of signal energy and several entropies to changes in skeletal muscle architecture in experimental muscular dystrophy before and after pharmacological therapeutic intervention [M. S. Hughes, IEEE Trans. UFFC. 54, 2291–2299 (2007)]. The study was based on a moving window analysis of simple cubic splines that were fit to the backscattered ultrasound and required that the radio frequency ultrasound (RF) be highly oversampled. The current study employs optimal smoothing splines instead to determine the effect of analyzing the same data with increasing levels of decimation. The RF data were obtained from selected skeletal muscles of muscular dystrophy mice (mdx: dystrophin -/-) that were randomly blocked into two groups: 4 receiving steroid treatment over 2 weeks, and 4 untreated positive controls. Ultrasonic imaging was performed on day 15. All mice were anesthetized then each forelimb was imaged in transverse cross sections using a Vevo-660 with a single-element 40 MHz wobbler-transducer (model RMV-704, Visualsonics). The result of each scan was a three dimensional data set $384 \times 8192 \times$ # frames in size. We find the equivalent sensitivity of this new approach for detecting treatment benefits as before ($p<0.03$), but now at a decimated sampling rate slightly below the Nyquist frequency. This implies that optimal smoothing splines are useful for analysis of data acquired from point of care imaging devices where hardware cost and power consumption must be minimized.

9:15

4aSPa2. Waveform processing using entropy instead of energy: A quantitative comparison based on the heat equation. Michael Hughes (Int. Med./Cardiology, Washington Univ. School of Medicine, 1632 Ridge Bend Dr., St Louis, MO 63108, mshatctrain@gmail.com), John McCarthy (Mathematics, Washington Univ., St Louis, MO), Jon Marsh (Int. Med./Cardiology, Washington Univ. School of Medicine, Saint Louis, MO), and Samuel Wickline (Mathematics, Washington Univ., Saint Louis, MO)

Virtually all modern imaging devices function by collecting electromagnetic or acoustic backscattered waves and using the energy carried by these waves to determine pixel values that build up what is basically an “energy” picture. However, waves also carry “information” that also may be used to compute the pixel values in an image. We have employed several measures of information, most sensitive being the “joint entropy” of the backscattered wave and a reference signal. Numerous published studies have demonstrated the advantages of “information imaging,” over conventional methods for materials characterization and medical imaging. A typical study is comprised of repeated acquisition of backscattered waves from a specimen that is changing slowing with acquisition time or location. The sensitivity of repeated experimental observations of such a slowly changing quantity may be defined as the mean variation (i.e., observed change) divided by mean variance (i.e., observed noise). Assuming the noise is Gaussian and using Wiener integration to compute the required mean values and variances, solutions to the Heat equation may be used to express the sensitivity for joint entropy and signal energy measurements. There always exists a reference such that joint entropy has larger variation and smaller variance than the corresponding quantities for signal energy, matching observations of several studies. A general prescription for finding an “optimal” reference for the joint entropy emerges, which has been validated in several studies.

9:30

4aSPa3. The classification of underwater acoustic target signals based on wave structure and support vector machine. Qingxin Meng, Shie Yang, and Shengchun Piao (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., No.145,Nantong St.,Nangang District, Harbin City, Heilongjiang Province 150001, China, mengqingxin005@hrbeu.edu.cn)

The sound of propeller is a remarkable feature of ship-radiated noise, the loudness and timbre of which are usually applied to identify types of ships. Since the information of loudness and timbre is indicated in the wave structure of time series, the feature of wave structure can be extracted to classify types of various underwater acoustic targets. In this paper, the method of feature vector extraction of underwater acoustic signals based on wave structure is studied. The nine-dimension features are constructed via signal statistical characteristics of zero-crossing wavelength, peek-to-peek amplitude, zero-crossing wavelength difference, and wave train areas. And then, the support vector machine (SVM) is applied as a classifier for two kinds of underwater acoustic target signals. The kernel function is set radial basis function (RBF). By properly setting the penalty factor and parameter of RBF, the recognition rate reaches over 89.5%, respectively. The sea-test data shows the validity of target recognition ability of the method above.

9:45

4aSPa4. Determination of Room Impulse Response for synthetic data acquisition and ASR testing. Philippe Moquin (Microsoft, One Microsoft Way, Redmond, WA 98052, pmoquin@microsoft.com), Kevin Venalainen (Univ. of Br. Columbia, Vancouver, BC, Canada), and Dinei A. Florêncio (Microsoft, Redmond, WA)

Automatic Speech Recognition (ASR) works best when the speech signal best matches the ones used for training. Training, however, may require thousands of hours of speech, and it is impractical to directly acquire them in a realistic scenario. Some improvement can be obtained by incorporating typical building acoustics measurement parameters such as RT, Cx, LF, etc., with limited gain. Instead, we estimate Room Impulse Responses (RIRs), and convolve speech and noise signals with the estimated RIRs. This produces realistic signals, which can then be processed by the audio pipeline, and used for ASR training. In our research, we use rooms with variable acoustics and repeatable source-receiver positions. The receivers are microphone arrays making the relative phase and magnitude critical. A standard mouth simulator for voice signals at various positions in the room is under robot control. A limited corpus of speech data as well as noise sources is recorded and the RIR at these 27 positions is determined using a variety of methods (chirp, MLS, impulse, and noise). The convolved RIR with the "clean speech" is compared to the actual measurements. Test methods used, differences from the measurements, and the difficulty of determining the unique RIR will be presented.

THURSDAY MORNING, 30 OCTOBER 2014 INDIANA G, 10:15 A.M. TO 12:00 NOON

Session 4aSPb

Signal Processing in Acoustics: Beamforming, Spectral Estimation, and Sonar Design

Brian E. Anderson, Cochair
Geophysics Group, Los Alamos National Laboratory, MS D443, Los Alamos, NM 87545

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

Contributed Papers

10:15

4aSPb1. Quantifying the depth profile of time reversal focusing in elastic media. Brian E. Anderson, Marcel C. Remillieux, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, bea@lanl.gov)

A focus of elastic energy on the surface of a solid sample can be useful to nondestructively evaluate whether the surface or the near-surficial region is damaged. Time reversal techniques allow one to focus energy in this manner. In order to quantify the degree to which a time reversal focus can probe near-surficial features, the depth profile of a time reversal focus must be quantified. This presentation will discuss numerical modeling and experimental results used to quantify the depth profile. [This work was supported by the U.S. Dept. of Energy, Fuel Cycle R&D, Used Fuel Disposition (Storage) Campaign.]

10:30

4aSPb2. Competitive algorithm blending for enhanced source separation of convolutive speech mixtures. Keith Gilbert (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 36 Walnut St., Berlin, MA 01503, kgilbert@umassd.edu), Karen Payton (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, N. Dartmouth, MA), Richard Goldhor, and Joel MacAuslan (Speech Technol. & Appl. Res., Corp., Bedford, MA)

This work investigates an adaptive filter network in which multiple blind source separation methods are run in parallel, and the individual outputs are combined to produce estimates of acoustic sources. Each individual algorithm makes assumptions about the environment (dimensions of enclosure, reflections, reverberation, etc.) and the sources (speech, interfering noise, position, etc.), which constitutes an individual hypothesis about the observed microphone outputs. The goal of this competitive algorithm blending (CAB) approach is to achieve the performance of the "true" method, i.e., the method that has full knowledge of the environment's and the sources' characteristics *a priori*, without any prior information. Results are given for time-invariant, critically- and over- determined, convolutive mixtures of speech and interfering noise sources, and the performance of the CAB method is compared with the "true" method in both the transient adaptation phase and in steady state.

10:45

4aSPb3. Structural infrasound signals in an urban environment. Sarah McComas, Henry Diaz-Alvarez, Mike Pace, and Mihan McKenna (US Army Engineer Res. and Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, sarah.mccomas@usace.army.mil)

Historically, infrasound arrays have been deployed in rural environments where anthropological noise sources are limited. As interest in monitoring sources at local distances grows in the infrasound community, it will be vital to understand how to monitor infrasound sources in an urban environment. Arrays deployed in urban centers have to overcome the decreased signal to noise ratio and reduced amount of real estate available to deploy an array. To advance the understanding of monitoring infrasound sources in urban environments, we deployed local and regional infrasound arrays on building rooftops of the campus of Southern Methodist University (SMU) and collected data for one seasonal cycle. The data was evaluated for structural source signals (continuous-wave packets) and when a signal was identified the back azimuth to the source was determined through frequency wavenumber analysis. This information was used to identify hypothesized structural sources; these sources were verified through direct measurement, structural numerical modeling and/or full waveform propagation modeling. Permission to publish was granted by Director, Geotechnical & Structures Laboratory.

11:00

4aSPb4. Design of a speaker array system based on adaptive time reversal method. Gee-Pinn J. Too, Yi-Tong Chen, and Shen-Jer Lin (Dept. of Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd., Tainan 701, Taiwan, z8008070@email.ncku.edu.tw)

A system for focusing sound around desired locations by using a speaker array of controlled sources is proposed. To increase acoustic signal in certain locations where the user is within but to reduce it in the other certain locations by controlling source signals is the main objective of this study. Based on adaptive time reversal theory, input weighting coefficients for speakers are evaluated for the speaker sources. Experiments and simulations with a speaker array of controlled sources are established in order to observe the distribution of sound field under different boundary and control conditions. The results indicate that based on the current algorithm, the difference of sound pressure level between bright point and dark point can be as high as 12 dB with an eight speakers array system.

11:15

4aSPb5. Focusing the acoustic signal of a maneuvering rotorcraft. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

An algorithm was developed and tested to blindly focus the acoustic spectra of a rotorcraft that was blurred by time-varying Doppler shifts and other effects such atmospheric distortion. First, the fundamental frequency generated by the main rotor blades of a rotorcraft was tracked using a fix-lag smoother. Then, the frequency estimates were used to resample the data in time using interpolation. Next, the motion compensated data were further focused using a technique based upon the phase gradient autofocus algorithm. The performance of the focusing algorithm was evaluated by analyzing the increase in the amplitude of the harmonics. For most of the data, the algorithm focused the harmonics between approximately 10–90 Hz to within 1–2 dB of an estimated upper bound obtained from conservation of energy and estimates of the Doppler shift. In addition, the algorithm was able to separate two closely spaced frequencies in the spectra of the rotorcraft. The algorithm developed can be used to preprocess data for classification, nulling, and tracking algorithms.

11:30

4aSPb6. Representing the structure of underwater acoustic communication data using probabilistic graphical models. Atulya Yellepeddi (Elec. Engineering/Appl. Ocean Phys. and Eng., Massachusetts Inst. of Technology/Woods Hole Oceanographic Inst., 77 Massachusetts Ave., Bldg. 36-683, Cambridge, MA 02139, atulya@mit.edu) and James C. Preisig (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Exploiting the structure in the output of the underwater acoustic communication channel in order to improve the performance of the communication system is a problem that has received much recent interest. Methods such as physical constraints and sparsity have been used to represent such structure in the past. In this work, we consider representing the structure of the received signal using probabilistic graphical models (more specifically Markov random fields), which capture the conditional dependencies amongst a collection of random variables. In the frequency domain, the inverse covariance matrix of the received signal is shown to have a sparse structure. Under the assumption that the signal may be modeled as a multivariate Gaussian random variable, this corresponds to a Markov random field. It is argued that the underlying cause of the structure is the cyclostationary nature of the signal. In practice, the received signal is not exactly cyclostationary, but data from the SPACE08 acoustic communication experiment is used to demonstrate that field data exhibits exploitable structure. Finally, techniques to exploit graphical model structure to improve the performance of wireless underwater acoustic communication are briefly considered.

11:45

4aSPb7. Choice of acoustics signals family in multi-users environment. Benjamin Ollivier, Frédéric Maussang, and René Garello (ITI, Institut Mines-Telecom / Telecom Bretagne - Lab-STICC, 655 Ave. du Technopole, Plouzané 29200, France, benjamin.ollivier@telecom-bretagne.eu)

Our application concerns a system immersed in an underwater acoustical context, with Nt transmitters and Nr slowly moving receivers. The objective is that all receivers detect the transmitted signals, in order to estimate the time of arrival (TOA) and then to facilitate the localization when several TOA (more than 3) are present. We have to choose a method to generate a number Ns of broad-band signals to use the Code Division Multiple Access (CDMA) modulation, specially adapted to our problem. This work is devoted to selecting Nt signals among the Ns available. The aim is to choose the most distinctly detectable ones. First, in a no Doppler context, the criterion of signals selection is based on a ratio between maximum of auto-correlation and cross-correlation. Second, in the presence of Doppler, we rely on Ambiguity Function which allows representing the correlation function to several frequency Doppler shifts. The choice of Nt signals is then based on ratio between maximum of auto-ambiguity and cross-ambiguity. In this paper we will highlight the relevance of the criteria (correlation, ambiguity function) in the choice of the most appropriate signals in function of the multi-users context.

Session 4aUW

Underwater Acoustics: Shallow Water Reverberation II

Brian T. Hefner, Chair
Applied Physics Laboratory, University of Washington, Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—8:00

Contributed Paper

8:05

4aUW1. SONAR Equation perspective on TREX13 measurements. Dajun Tang and Brian T. Hefner (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Modeling shallow water reverberation is a problem that can be approximated as two-way propagation (including multiple forward scatter) and a single backward scatter. This can be effectively expressed in terms of the SONAR equation: RL = SL-2 x TL + SS, where RL is reverberation level, SL is the source level, TL is the one way transmission loss, and SS is the integrated scattering strength. In order to understand the reverberation problem at the basic research level, both propagation and scattering physics need to be properly addressed. A major goal of TREX13 (Target and Reverberation EXperiment 2013) is to quantitatively investigate reverberation with sufficient environmental measurement to support full modeling of reverberation data. Along a particular reverberation track at the TREX13 site, TL and direct-path backscatter were separately measured. Environmental data were extensively collected along this track. This talk will bring together all the components of the SONAR equation measured separately at the TREX13 site to provide an assessment of the reverberation process along with environmental factors impacting each of the components.

Invited Papers

8:20

4aUW2. Environmental measurements collected during TREX13 to support acoustic modeling. Brian T. Hefner and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Appl. Phys. Lab., University of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

The major goal of TREX13 (Target and Reverberation EXperiment 2013) was to quantitatively investigate reverberation with sufficient environmental measurements to support full modeling of reverberation data. The collection of environmental data to support reverberation modeling is usually limited by the large ranges (10s of km) involved, the temporal and spatial variability of the environment and the time variation of towed source/receiver locations within this environment. In order to overcome these difficulties, TREX13 was carried out in a 20 m deep shelf environment using horizontal line arrays mounted on the seafloor. The water depth and well controlled array geometry allowed environmental characterization to be focused on the main beam of the array, i.e., along a track roughly 5 km long and 500 m wide. This talk presents an overview of the efforts made to characterize the sea surface, water column, seafloor, and sub-bottom along this track to support the modeling of acoustic data collected over the course of the experiment. [Work supported by ONR Ocean Acoustics.]

8:40

4aUW3. Persistence of sharp acoustic backscatter transitions observed in repeat 400 kHz multibeam echosounder surveys offshore Panama City, Florida, over 1 and 24 months. Christian de Moustier (10dBx LLC, PO Box 81777, San Diego, CA 92138, cpm@ieee.org) and Barbara J. Kraft (10dBx LLC, Barrington, New Hampshire)

The Target and Reverberation Experiment 2013 (TREX13), conducted offshore Panama City, FL, from April to June 2013, sought to determine which environmental parameters contribute the most to acoustic reverberation and control sonar performance prediction modeling for acoustic frequencies between 1 kHz and 10 kHz. In that context, a multibeam echosounder operated at 400 kHz was used to map the seafloor relief and its high-frequency acoustic backscatter characteristics along the acoustic propagation path of the reverberation experiment. Repeat surveys were conducted a month apart, before and after the main reverberation experiment. In addition, repeat surveys were conducted at 200 kHz in April 2014. Similar mapping work was also conducted in April 2011 during a pilot experiment (GulfEx11) near the site chosen for TREX13. Both experiments revealed a persistent occurrence of sharp transitions from high to low acoustic backscatter at the bottom of swales. Hypotheses are presented for observable differences in bathymetry and acoustic backscatter in the overlap region between the GulfEx11 survey and the TREX13 surveys conducted 2 y apart. [Work supported by ONR 322 OA.]

Contributed Papers

9:00

4aUW4. Roughness measurement by laser profiler and acoustic scattering strength of a sandy bottom. Nicholas P. Chotiros, Marcia J. Isakson, Oscar E. Siliceo, and Paul M. Abkowitz (Appl. Res. Labs., Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, chotiros@arlut.utexas.edu)

The roughness of a sandy seabed off Panama City, FL, was measured with a laser profiler. This was the site of the target and reverberation experiment of 2013 (TREX13) in which propagation loss and reverberation strength were measured. The area may be characterized as having small scale roughness due to bioturbation overlaying larger sand ripples due to current activity. The area was largely composed of sand with shell hash crossed by ribbons of softer sediment at regular intervals. The roughness measurements were concentrated in the areas where the ribbons intersected the designated sound propagation track. Laser lines projected on the sand were imaged by a high-definition video recorder. The video images were processed to yield bottom profiles in three dimensions. Finally, the roughness data are used to estimate acoustic bottom scattering strength. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

9:15

4aUW5. Seafloor sub-bottom Imaging along the TREX reverberation track. Joseph L. Lopes, Rodolf Arrieta, Iris Paustian, Nick Pineda (NSWC PCD, 110 Vernon Ave, Panama City, FL 32407-7001, joseph.l.lopes@navy.mil), and Kevin Williams (Appl. Phys. Lab. / Univ. of Washington, Seattle, WA)

The Buried Object Scanning Sonar (BOSS) integrated into a Bluefin12 autonomous underwater vehicle was used to collect seafloor sub-bottom data along the TREX reverberation track. BOSS is a downward looking sonar and employs an omni-directional source to transmit a 3 to 20 kHz linear frequency modulated (LFM) pulse. Backscattered signals are received by two 20-channel linear hydrophone arrays. The BOSS survey was carried out to support long-range reverberation measurements at 3 kHz. The data were beamformed in three dimensions and processed into 10cm x 10cm x 10cm voxel maps of backscattering to a depth of 1 m. Comparison of the BOSS imagery with 400 kHz multibeam sonar imagery of the seafloor allows muddy regions to be identified and shows differences rationalized by the differences in sediment penetration of the two frequency ranges utilized. Processed BOSS data are consistent with observations from diver cores and the reverberation data collected by stationary arrays deployed on the seafloor. Specifically, stronger and deeper backscattering from muddy regions is observed (relative to near-by sandy regions). This correlates well with the large amounts of detritus (e.g., shell fragments) and complicated vertical layering within cores, and the enhanced reverberation, from those regions. [Work supported by ONR.]

9:30

4aUW6. Seabed characterisation using a low cost digital thin line array: Results from the Target and Reverberation Experiments 2013. Unnikrishnan K. Chandrika, Venugopalan Pallayil (Acoust. Res. Lab, TMSI, National Univ. of Singapore, Acoust. Res. Lab, 18 Kent Ridge Rd., Singapore 119227, Singapore, venu@arl.nus.edu.sg), Nicholas Chotiros (Appl. Res. Lab, Univ. of Texas, Austin, TX), and Marcia Isakson (Appl. Res. Lab, Univ. of Texas, Ausitn, TX)

During TREX-13 experiments in the Gulf of Mexico in May 2013, the use of a low cost digital thin line array (DTLA) developed at the Acoustic Research Lab, National University of Singapore was explored towards seabottom characterisation. The array, developed for use from AUV platforms, was hosted on a Sea-eye ROV from UT Austin and towed using R/V Smith, as no AUV platform was available during the course of the above experiment. The ROV was also hosting a wide-band acoustic source sending out chirp waveforms in the frequency range of 3 to 15 kHz. It has been observed that despite the complexity of set-up used, the array dynamics could be maintained well during the tow test and also the data collected were useful in estimating the bottom type from reflection coefficient measurements and comparing with the models available. Our analysis by matched filtering the received data and estimating the bottom reflection coefficient showed that the bottom type at the experimental site was sandy-silt, which fairly compared with observations on the same by other means. Details of experiments performed and the results from the data analyzed would be presented during the meeting. Some suggestions for improvement for future experiments will be discussed.

9:45

4aUW7. Wide-angle reflection measurements (TREX13): Evidence of strong seabed lateral heterogeneity at two scales. Charles W. Holland, Chad Smith (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Paul Hines (Elec. and Comput. Eng., Dalhousie Univ., Dalhousie, NS, Canada), Jan Dettmer, Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Samuel Pinson (Appl. Res. Lab., The Penn State Univ., State College, PA)

Broadband wide-angle reflection data possess high information content, yielding both depth and frequency dependence of sediment wave velocities, attenuations, and density. Measurements at two locations off Panama City, FL (TREX13), however, presented a surprise: over the measurement aperture (a few tens of meters) the sediment was strongly laterally variable. This prevented the usual analysis in terms of depth dependent geoacoustic properties. Only rough estimates could be made. On the other hand, the data provide clear evidence of lateral heterogeneity at O(10^0-10^1) m scale. The two sites were separated by ~6 km, one on a ridge (lateral dimension 10^2 m) and one in a swale of comparable dimension; the respective sound speeds are roughly 1680 m/s and 1585 m/s. The lateral variability, especially at the 1–10 m scale is expected to impact both propagation and reverberation. Characteristics of the reflection data and its attendant “surprise” suggest the possibility of objectively separating the intermingled angle and range dependence; this would open the door to detailed geoacoustic estimation in areas of strong lateral variability. [Research supported by ONR Ocean Acoustics.]

Invited Papers

10:00

4aUW8. Modeling reverberation in a complex environment with the finite element method. Marcia J. Isakson and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Acoustic finite element models solve the Helmholtz equation exactly and are customizable to the scale of the discretization of the environment. This makes them an ideal candidate for reverberation studies in complex environments. In this study, reverberation is calculated for a realistic shallow water waveguide. The environmental parameters are taken from the extensive characterization completed for the Target and Reverberation Experiment (TREX) conducted off the coast of the Florida panhandle in 2013. Measured sound speed profiles, sea surface roughness, bathymetry, and measured ocean bottom roughness are included in the model. Measurements of the normal incidence bottom loss are used as a proxy for range dependent sediment density. Results are compared with a closed form solution for reverberation. [Work sponsored by ONR, Ocean Acoustics.]

10:20–10:35 Break

Contributed Papers

10:35

4aUW9. Normal incidence reflection measurements (TREX13): Inferences for lateral heterogeneity over a range of scales. Charles W. Holland, Chad Smith (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), and Paul Hines (Elec. and Comput. Eng., Dalhousie Univ., Dalhousie, NS, Canada)

Normal incidence seabed reflection data suffer from a variety of ambiguities that make quantitative interpretation difficult. The reflection coefficient has an inseparable ambiguity between bulk density and compressional sound speed. Even more serious, reflection data are a function of other sediment characteristics including interface roughness, volume heterogeneities, and local bathymetry. Seafloor interface curvature is especially important and can lead to focusing/defocusing of the reflected field. An attempt is made with ancillary data including bathymetry, 400 kHz backscatter, and wide angle seabed reflection data to separate some of the mechanisms. Resulting analysis of 1–12 kHz reflection data suggest: (1) strong lateral sediment heterogeneity exists on scales of 10–100 m; (2) there are distinct geoacoustic regimes on the lee and stoss side of the ridge crest, and also between crest and the swale, and (3) the ridge crest geoacoustic properties are similar across distances of 6 km along two perpendicular transects (1 correlation). [Research supported by ONR Ocean Acoustics.]

10:50

4aUW10. Acoustic measurements on mid-shelf sediments with cobble: Implications for reverberation. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Gavin Steininger, Jan Dettmer, Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Allen Lowrie (Picayune, MS)

The vast majority of sediment acoustics research has focused on rather homogeneous sandy sediments. Measurements for sediments containing cobbles (grain size greater than 6 cm) are rare. Here, measurements are presented for mid-shelf sediments containing pebbles/cobbles mixed with other grain sizes spanning 7 orders of magnitude, including silty clay, sand, and shell hash. The 2 kHz sediment sound speed in two distinct layers with cobble is 1531 ± 5 m/s and 1800 ± 20 m/s at the 95% credibility interval. The dispersion over the 400–2000 Hz band was relatively weak, 2 and 7 m/s respectively. The objective is to (1) present results for a sediment type for which little is known, (2) motivate development of theoretical wave propagation models for wide grain size distributions, and (3) speculate on the possibility of cobble as a scattering mechanism for mid shelf reverberation. The presence of cobbles from 1 to 3 m (possibly extending to 6 m) sub-bottom suggest they are the dominant scattering mechanism at this site. Though sediments with cobbles might be considered unusual, especially on the mid-shelf, they may be more common than the paucity of measurements would suggest since typical direct sampling techniques (e.g., cores and grab samples) have fundamental sampling limitations. [Research supported by ONR Ocean Acoustics.]

Session 4pAAa

Architectural Acoustics and Speech Communication: Acoustic Trick-or-Treat: Eerie Noises, Spooky Speech, and Creative Masking

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Eric J. Hunter, Cochair
Department of Communicative Sci., Michigan State University, 1026 Red Cedar Road, East Lansing, MI 48824

Chair's Introduction—1:10

Invited Papers

1:15

4pAAa1. Auditory illusions of supernatural spirits: Archaeological evidence and experimental results. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com) and Miriam A. Kolar (Amherst College, Amherst, MA 01002)

Sound reflection, reverberation, ricochets, and interference patterns were perceived in the past as eerie sounds attributable to invisible echo spirits, thunder gods, ghosts, and sound-absorbing bodies. These beliefs in the supernatural were recorded in ancient myths, and expressed in tangible archaeological evidence including canyon petroglyphs, cave paintings, and megalithic stone circles including Stonehenge. Disembodied voices echoing throughout canyons gave the impression of echo spirits calling out from the rocks. Thunderous reverberation filling deep caves gave the impression of the same thundering stampedes of invisible hoofed animals that were believed to accompany thunder gods in stormy skies. If you did not know about sound wave reflection, would the inexplicable noise of a ricochet in a large room have given you the impression of a ghost moaning "BOOoo" over your shoulder? Mysterious silent zones in an open field gave the impression of a ring of large phantom objects blocking pipers' music. Complex behaviors of sound waves such as reflection and interference (which scientists today dismiss as acoustical artifacts) can experimentally give rise to psychoacoustic misperceptions in which such unseen sonic phenomena are attributed to the supernatural. See https://sites.google.com/site/rockartacoustics/ for further details.

1:35

4pAAa2. Pututus, resonance and beats: Acoustic wave interference effects at Ancient Chavín de Huántar, Perú. Miriam A. Kolar (Program in Architectural Studies and Dept. of Music, Amherst College, Barrett Hall, 21 Barrett Hill Dr., AC# 2255, PO Box 5000, Amherst, MA 01002, mkolar@amherst.edu)

Acoustic wave interference produces audible effects observed and measured in archaeoacoustic research at the 3,000-year-old Andean Formative site at Chavín de Huántar, Perú. The ceremonial center's highly-coupled network of labyrinthine interior spaces is riddled with resonances excited by the lower-frequency range of site-excavated conch shell horns. These *pututus*, when played together in near-unison tones, produce a distinct "beat" effect heard as the result of the amplitude variation that characterizes this linear interaction. Despite the straightforward acoustic explanation for this architecturally enhanced instrumental sound effect, the performative act reveals an intriguing perceptual complication. While playing pututus inside Chavín's substantially intact stone-and-earthen-mortar buildings, pututu performers have reported an experience of having their instruments' tones "guided" or "pulled" into tune with the dominant spatial resonances of particular locations. In an ancient ritual context, the recognition and understanding of such a sensory component would relate to a particular worldview beyond the reach of present-day investigators. Despite our temporal distance, an examination of the intertwined acoustic phenomena operative to this architectural–instrumental–experiential puzzle enriches the interdisciplinary research perspective, and substantiates perceptual claims.

1:55

4pAAa3. Tapping into the theatre of the mind; creating the eerie scene through sound. Jonathon Whiting (Media and Information, Michigan State Univ., College of Commun. Arts and Sci., 404 Wilson Rd., Rm. 409, East Lansing, MI 48824, whitin26@msu.edu)

Jaws. Psycho. Halloween. Halo. Movies and video games depend on music and acoustics to evoke certain emotional states in the audience or game player. But what is the recipe for creating a haunting scene? A creaky door, a scream, a minor chord on a piano. How and why are certain emotions pulled out of a listener in response to sound? From sound environments to mental expectations, the media industry uses a variety of techniques to elicit responses from an audience. This presentation will discuss and present examples of the principles behind the sound of fright.

2:15

4pAAa4. Disquiet: Epistemological bogeymen and other exploits in audition. Ean White (unaffiliated, 1 Westinghouse Plaza C-216, Boston, MA 02136-2079, ean@eanwhite.org)

Beginning with an interest in "physiological musics," Ean White's sound art exploits interstices in our sensory apparatus with unnerving results. He will recount a series of audio experiments with effects ranging from involuntary muscle contractions to the creation of sounds eerily unique to each listener. The presentation will include discussion of his techniques and how they inform his artistic practice.

2:35

4pAAa5. Removing the mask in multitrack music mixing. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The sound recording heard via stereo loudspeakers and headphones is made up of many dozens—sometimes more than 100—discrete tracks of musical elements. Multiple individual performances across a variety of instruments are fused into the final, two-channel recording—left and right—that is released to consumers. Achieving sonic success in this many-into-two challenge requires strategic, creative release from masking. Part of the artistry of multitrack mixing includes finding innovative signal processing approaches that enable the full arrangement and the associated interaction among the multitrack components of the music to be heard and enjoyed. Masking among tracks clutters and obscures the music. But audio engineers are not afraid. They want you hear what's behind the mask. Hear how. Happy Halloween.

2:55

4pAAa6. Documenting and identifying things that go bump in the night. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

Acoustical consultants are occasionally asked to help diagnose mysterious noises in buildings, and it can be difficult to be present and ready to make measurements when such noises occur. This paper will presents some of the tools and methods the author uses for recording and analyzing these events. These include the use of tablet-based measurement devices and high-speed playback of long-term recordings.

3:15–3:30 Break

3:30

4pAAa7. Inaudible information, disappearing declamations, misattributed locations, and other spooky ways your brain fools you—every day. Barbara Shinn-Cunningham (Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

We bumble through life convinced that our senses provide reliable, faithful information about the world. Yet on closer inspection, our brains constantly misinform us, creepily convincing us of "truths" that are just plain false. We hear information that is not really there. We are oblivious to sounds that are perfectly audible. For sounds that we do hear, we cannot tell when they actually occurred. We completely overlook changes that even a simple acoustic analysis would detect with 100% accuracy. In short, we misinterpret the sounds reaching our ears all the time, and do not even realize it. This talk will review the evidence for how unreliable and biased we are in interpreting the world—and why the chilling failures of our perceptual machinery may be excusable, or even useful, as we navigate the complex world in which we live.

3:50

4pAAa8. The mysterious case of the singing toilets and other nerve wracking tales of unwanted sound. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

Lightweight construction nightmares, devilish designs that never see acoustic review, improper purposing of spaces, and other stories involving the relentless torture of building occupants. Will they survive?

4:10

4pAAa9. Sound effects with AUditory syntaX—A high-level scripting language for sound processing. Bomjun J. Kwon (Hearing, Speech and Lang., Gallaudet University, 800 Florida Ave NE, Washington, DC 20002, bomjun.kwon@gallaudet.edu)

AUditory syntaX (AUX) is a high-level scripting programming language specifically crafted for the generation and processing of auditory signals (Kwon, 2012; Behav Rev 44, 361–373). AUX does not require knowledge or prior experience in computer programming. Rather, AUX provides an intuitive and descriptive environment where users focus on perceptual components of the sound, without tedious tasks unrelated to the perception such as memory management or array handling often required in other computer languages such as C++ or MATLAB that are popularly used in auditory science. This presentation provides a demonstration of AUX for the generation and processing of various sound effects, particularly "fun" or "spooky" sounds. Processing methods for sound effects widely used in arts, films and other media, such as reverberation, echoes, modulation, pitch shift, and flanger/phaser, will be reviewed and coding in AUX to generate those effects and the generated sounds will be demonstrated.

4:30

4pAAa10. Eerie voices: Odd combinations, extremes, and irregularities. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

The human voice can project an eerie quality when certain characteristics are present in a particular context. Some types of eerie voices may be derived from physiological scaling of the speech production system that is either humanly impossible or nearly so. By combining previous work on adult speech, and current research on speech development, the purpose of this study was to simulate vocalizations and speech based on unusual configurations of the vocal tract and vocal folds, and by imposing irregularities on movement and vibration. The resulting sound contains qualities that are human-like, but not typical, and hence may give the perceptual impression of eeriness. [Supported in part by NIH R01-DC011275.]

4:50

4pAAa11. Segregation of ambiguous pulse-echo streams and suppression of clutter masking in FM bat sonar by anticorrelation signal processing. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu)

Big brown bats often fly in conditions where the density and spatial extent of clutter requires a high rate of pulse emissions. Echoes from one broadcast still are arriving when the next broadcast is sent out, creating ambiguity about matching echoes to corresponding broadcasts. Biosonar sounds are widely beamed and impinge on the entire surrounding scene. Numerous clutter echoes typically are received from different directions at similar times. The multitude of overlapping echoes and the occurrence of pulse-to-echo ambiguity compromises the bat's ability to peer into the upcoming path and determine whether it is free of collision hazards. Bats have to associate echoes with their corresponding broadcasts to prevent ambiguity, and off-side clutter echoes have to be segregated from on-axis echoes that inform the bat about its immediate forward path. In general, auditory streaming to resolve elements of an auditory scene depends on differences in pitch and temporal pattern. Bats use a combination of temporal and spectral pitch to assign echoes to "target" and "clutter" categories within the scene, which prevents clutter masking, and they associate incoming echoes with the corresponding broadcast by treating the mismatch of echoes with the wrong broadcast as a type of clutter. [Supported by ONR.]

5:10

4pAAa12. Are you hearing voices in the high frequencies of human speech and voice? Brian B. Monson (Pediatric Newborn Medicine, Brigham and Women's Hospital, Harvard Med. School, 75 Francis St., Boston, MA 02115, bmonson@research.bwh.harvard.edu)

The human voice produces acoustic energy at frequencies above 6 kHz. Energy in this high-frequency region has long been known to affect perception of speech and voice quality, but also provides non-qualitative information about a speech signal. This presentation will demonstrate how much useful information can be gleaned from the high frequencies with a report on studies where listeners were presented with only high-frequency energy extracted from speech and singing. Come to test your own abilities and decide if you can hear strange voices or just chirps and whistles in the high frequencies of human speech and voice.

Contributed Paper

5:30

4pAAa13. Measuring the impact of room acoustics on emotional responses to music using functional neuroimaging: A pilot study. Martin S. Lawless and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu)

Past cognitive neuroscience studies have established links between music and an individual's emotional response. Specifically, music can induce activations in brain regions most commonly associated with reward and pleasure (Blood/Zatorre PNAS 2001). To further develop concert hall design criteria, functional magnetic resonance imaging (fMRI) techniques can be used to investigate the emotional preferences of room acoustics stimuli. Auralizations were created under various settings ranging from anechoic to extremely reverberant. These stimuli were presented to five participants in an MRI machine, and the subjects were prompted to rate the stimuli in terms of preference. Noise stimuli that matched the acoustic stimuli temporally and spectrally were also presented to the participants for the analysis of main contrasts of interest. In addition, the participants were first tested in a mock scanner to acclimatize the subjects to the environment and later validate the results of the study. Voxel-wise region of interest analysis was used to locate the emotion and reward epicenters of the brain that were activated when the subjects enjoyed a hall's acoustics. The activation levels of these regions, which are associated with positive-valence emotions, were examined to determine if the activations correlate with preference ratings.

Session 4pAAb

Architectural Acoustics, Speech Communication, and Noise: Room Acoustics Effects on Speech Comprehension and Recall II

Lily M. Wang, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

David H. Griesinger, Cochair
Research, David Griesinger Acoustics, 221 Mt Auburn St #504, Cambridge, MA 02138

Invited Papers

1:15

4pAAb1. Challenges for second-language learners in difficult acoustic environments. Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

Most anyone who has lived in a foreign country for any length of time knows that even everyday tasks can become tiring and frustrating when one must accomplish them while navigating a seemingly endless maze of unfamiliar social customs, vocabulary and speech that seem far removed from one's language laboratory experience. Add to these challenges noise, reverberation, and/or cognitive demand (e.g., learning caculus, responding to multiple customer, and co-worker demands) and even experienced learners may begin to question their proficiency. This presentation will provide an overview of the speech perception and production challenges faced by second-language learners in difficult acoustic environments that we may encounter every day, such as in large lecture halls, retail or customer service, to name a few. Past and current research investigating the effects of various environmental challenges on both relatively early and later learners of a second language will be considered, as well as strategies that may mitigate challenges for both speakers and listeners in some of these conditions.

1:35

4pAAb2. Development of speech perception under adverse listening conditions. Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

Speech communication success is dependent on interactions among the talker, listener, and listening environment. One such important interaction is between the listener's age and the noise and reverberation in the environment. Previous work has demonstrated that children have greater difficulty than adults in noisy and highly reverberant environments, such as those frequently found in classrooms. I will review research that considers how a talker's production patterns also contribute to speech comprehension, focusing on nonnative talkers. Studies from my lab have demonstrated that children have more difficulty than adults perceiving speech that deviates from native language norms, even in quiet listening conditions in which adults are highly accurate. When a nonnative talker's voice was combined with noise, children's word recognition was particularly poor. Therefore, similar to the developmental trajectory for speech perception in noise or reverberation, the ability to accurately perceive speech produced by nonnative talkers continues to develop well into childhood. Metrics to quantify speech intelligibility in specific rooms must consider both listener characteristics, talker characteristics, and their interaction. Future research should investigate how children's speech comprehension is influenced by the interaction between specific types of background noise and reverberation and talker production characteristics. [Work supported by NIH-R21DC010027.]

1:55

4pAAb3. Measurement and prediction of speech intelligibility in noise and reverberation for different sentence materials, speakers, and languages. Anna Warzybok, Sabine Hochmuth (Cluster of Excellence Hearing4All, Medical Phys. Group, Universität Oldenburg, Oldenburg D-26111, Germany, a.warzybok@uni-oldenburg.de), Jan Rennies (Cluster of Excellence Hearing4All, Project Group Hearing, Speech and Audio Technol., Fraunhofer Inst. for Digital Media Technol. IDMT, Oldenburg, Germany), Thomas Brand, and Birger Kollmeier (Cluster of Excellence Hearing4All, Medical Phys. Group, Universität Oldenburg, Oldenburg, Germany)

The present study investigates the role of the speech material type, speaker, and language for speech intelligibility in noise and reverberation. The experimental data are compared to predictions of the speech transmission index. First, the effect of noise only, reverberation only, and the combination of noise and reverberation was systematically investigated for two types of sentence tests. The hypothesis to be tested was that speech intelligibility is more affected by reverberation when using an open-set speech material consisting of everyday sentences than when using a closed-set test with syntactically fixed and semantically unpredictable sentences. In order to distinguish between the effect of speaker and language on speech intelligibility in noise and reverberation, the closed-set speech material was recorded using bilingual speakers of German-Spanish and German-Russian. The experimental data confirmed that the effect of

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reverberation was stronger for an open-set test than for a closed-set test. However, this cannot be predicted by the speech transmission index. Furthermore, the inter-language differences in speech reception thresholds were on average up to 5 dB, whereas inter-talker differences were of about 3 dB. The Spanish language suffered more under reverberation than German and Russian, what again challenged the predictions of the speech transmission index.

2:15

4pAAb4. Speech comprehension in realistic classrooms: Effects of room acoustics and foreign accent. Zhao Peng, Brenna N. Boyd, Kristin E. Hanna, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182, zpeng@huskers.unl.edu)

The current classroom acoustics standard (ANSI S12.60) recommends that core learning spaces shall not exceed reverberation time (RT) of 0.6 second and background noise level (BNL) of 35 dBA, based on speech intelligibility performance mainly by the native English-speaking population. This paper presents two studies on the effects of RT and BNL on more realistic classroom learning experiences. How do native and non-native English-speaking listeners perform on speech comprehension tasks under adverse acoustic conditions, if the English speech is produced by talkers whose native language is English (Study 1) versus Mandarin Chinese (Study 2)? Speech comprehension materials were played back in a listening chamber to individual listeners: native and non-native English-speaking in Study 1; native English, native Mandarin Chinese, and other non-native English-speaking in Study 2. Each listener was screened for baseline English proficiency for use as a covariate in the statistical analysis. Participants completed dual tasks simultaneously (speech comprehension and adaptive dot-tracing) under 15 different acoustic conditions, comprised of three BNL conditions (RC-30, 40, and 50) and five RT scenarios (0.4–1.2 s). Results do show distinct differences between the listening groups. [Work supported by a UNL Durham School Seed Grant and the Paul S. Veneklasen Research Foundation.]

Contributed Papers

2:35

4pAAb5. Speech clarity in lively theatres. Gregory A. Miller and Carl Giegold (Threshold Acoust., LLC, 53 W. Jackson Boulevard, Ste. 815, Chicago, IL 60604, gmiller@thresholdacoustics.com)

By their very nature, theatres must be "lively" acoustic spaces. The audience must hear one another, so laughter and applause can ripple around the room, and they must have the aural sensation of being in a large space heightens the excitement of being at a live performance. Similarly, the theatre must reflect sound back to the actors in a way that helps them to gauge how well their voices are filling the room, and to gauge audience response throughout the performance. And yet this liveliness runs counter to much of conventional wisdom regarding the acoustic conditions to support speech clarity. This paper will describe ways in which the acoustic response of a room can be built up to support both speech clarity and liveliness, with a particular emphasis on theatre spaces in which the actors are placed in the same volume as the audience (thrust and surround stages).

2:50

4pAAb6. Speech communication in noise to valid the virtual sound capturing system. Hyung Suk Jang, Seongmin Oh, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, South Korea, janghyungs@gmail.com)

The microphone systems were designed to capture the real sound field for the creation of the remote virtual coexistence space: omnidirectional microphone, binaural dummy head, linear array microphones, and spherical microphone. The captured signals were applied to synthesize into the binaural signal. These binaural cues were generated using head-related transfer function (HRTF) through headphone. For the validation, the sentence recognition tests were carried out to quantify the ability of speech perception with the sentence lists for normal listeners. In addition, the readability and the naturalness were used to assess the quality of the synthesized sounds. The different noise environments were applied with different signal to noise ratio and an efficient sound capturing system was suggested by the comparing the results of the sentence recognition tests.

3:05

4pAAb7. Quantifying a measure and exploring the effect of varying reflection densities from realistic room impulse responses. Hyun Hong and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, hhong@huskers.unl.edu)

Perceptual studies using objective acoustic metrics calculated from room impulse responses, such as reverberation time and clarity index, are common. Less work has been conducted looking explicitly at the reflection density, or the number of reflections per second. The reflection density, though, may well have its own perceptual influence when reverberation time and source-receiver distances are controlled, particularly in relation to room size perception. This paper presents first an investigation into quantifying the reflection density from realistic room impulse responses that may be measured or simulated. The resolution of the sampling frequency, time window applied, and cut-off level for including a reflection in the count are considered. The quantification method is subsequently applied to select a range of realistic RIRs for use in a perceptual study on determining the maximum audible reflection density by humans, using both speech and clapping signals. Results from this study are compared to those from similar previous work by the authors which used artificially simulated impulse responses with constant reflection densities over time.

Session 4pAB

Animal Bioacoustics and Acoustical Oceanography: Use of Passive Acoustics for Estimation of Animal Population Density II

Tina M. Yack, Cochair
Bio-Waves, Inc., 364 2nd Street, Suite #3, Encinitas, CA 92024

Danielle Harris, Cochair
Centre for Research into Ecological and Environmental Modelling, University of St. Andrews, The Observatory, Buchanan Gardens, St. Andrews KY16 9LZ, United Kingdom

Chair's Introduction—1:15

Invited Papers

1:20

4pAB1. Estimating singing fin whale population density using frequency band energy. David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Elizabeth T. Küsel (NW Electromagnetics and Acoust. Res. Lab., Portland State Univ., Portland, OR), Danielle Harris, Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), and Luis Matias (Instituto Dom Luiz, Faculdade de Ciências, Universidade de Lisboa, Lisbon, Portugal)

Fin whale (*Balaenoptera physalus*) song occurs in a narrow frequency band between approximately 15 and 25 Hz. During the breeding season, the sound from many distant fin whales in tropical and subtropical parts of the world may be seen as a "hump" in this band of the ocean acoustic spectrum. Since a higher density of singing whales leads to more energy in the band, the size of this hump—the total received acoustic energy in this frequency band—may be used to estimate the population density of singing fin whales in the vicinity of a sensor. To estimate density, a fixed density of singing whales is simulated; using acoustic propagation modeling, the energy they emit is propagated to the sensor, and the received level calculated. Since received energy in the fin whale band increases proportionally with the density of whales, the density of whales may then be estimated from the measured received energy. This method is applied to a case study of sound recorded on ocean-bottom recorders southwest of Portugal; issues covered include variance due to acoustic propagation modeling, reception area, variation in whale song acoustic level and frequency, and elimination of interfering sounds. [Funding from ONR.]

1:40

4pAB2. Large-scale passive-acoustics-based population estimation of African forest elephants. Yu Shiu, Sara Keen, Peter H. Wrege, and Elizabeth Rowland (BioAcoust. Res. Program, Cornell Univ., 159 Sapsucker Woods Rd, Ithaca, NY 14850, atoultaro@gmail.com)

African forest elephants (*Loxodonta cyclotis*) live in tropical rainforests in Central Africa and often use low-frequency vocalizations for long-distance communication and coordination of group activities. There is great interest in monitoring population size in this species; however, the dense rainforest canopy severely limits visibility, making it difficult to estimate abundance using traditional methods such as aerial surveys. Passive acoustic monitoring offers an alternative approach of estimating its abundance in a low visibility environment. The work we present here can be divided into three steps. First, we apply an automatic elephant call detector, which enables the processing of large-scale acoustic signals in a reasonable amount of time. Second, we apply a density estimation method we designed for a single microphone. Because microphones are often positioned far apart in order to cover a large area in the rainforest, meaning that the same call will not produce multiple arrivals on different recording units. Lastly, we examine results from our historic data across five years in six locations in central Africa, which includes over 1000 days of sound stream. We will address the feasibility of long-term population monitoring and also the potential impact of human activity on elephant calling behavior.

2:00

4pAB3. A generalized random encounter model for estimating animal density with remote sensor data. Elizabeth Moorcroft, Tim C. D. Lucas (Ctr. for Mathematics, Phys. and Eng. in the Life Sci. and Experimental Biology, UCL, CoMPLEX, University College London, Gower St., London WC1E 6BT, United Kingdom, e.moorcroft@ucl.ac.uk), Robin Freeman, Marcus J. Rowcliffe (Inst. of Zoology, Zoological Society of London, London, United Kingdom), and Kate E. Jones (Ctr. for Biodiversity and Environment Res., UCL, London, United Kingdom)

Acoustic detectors are commonly being used to monitor wildlife. Current estimators of abundance or density require recognition of individuals or the distance of the animal from the sensor, which is often difficult. The random encounter model (REM) has been successfully applied to count data without these requirements. However, count data from acoustic detectors do not fit the assumptions of the REM due to the directionality of animal signals. We developed a generalized REM (gREM), to estimate animal density from count data, derived for different combinations of sensor detection widths and animal signal widths. We tested the accuracy and precision of this model using simulations for different combinations of sensor detection and animal signal widths, number of captures, and animal movement models. The gREM produces accurate estimates of absolute animal density. However, larger sensor detection and animal signal widths, and larger number of captures give more precise estimates. Different animal movement models had no effect on the gREM. We conclude that the gREM provides an effective method to estimate animal densities in both marine and terrestrial environments. As acoustic detectors become more ubiquitous, the gREM will be increasingly useful for monitoring animal populations across broad spatial, temporal, and taxonomic scales.

2:20

4pAB4. Using sound propagation modeling to estimate the number of calling fish in an aggregation from single-hydrophone sound recordings. Mark W. Sprague (Phys., East Carolina Univ., M.S. 563, Greenville, NC 27858, spraguem@ecu.edu) and Joseph J. Luczkovich (Biology, East Carolina Univ., Greenville, NC)

Many fishes make sounds during spawning events that can be used to estimate abundance. Spawning stock size is a measure of fish population size that is used by fishery biologists to manage harvests levels. It is desirable that such an estimate be assessed easily and remotely using passive acoustics. Passive acoustics techniques (hydrophones) can be used to identify sound-producing species, but it is difficult to count individual sound sources in the sea, where it is dark, background noise levels can be high, but species can be identified by their sounds. We have developed a method that can estimate the density of calling fish in an aggregation from single-hydrophone recordings. Our method requires a sound propagation model for the area in which the aggregation is located. We generate a library of modeled sounds of virtual Monte-Carlo generated distributions of fish to determine the range of fish population densities that match the characteristics of single-hydrophone sound recording. Such a model could be used from a fixed station (e.g., an observatory) to estimate the population size of the sound producers. In this presentation, we will present some calculations made using this method and will examine the benefits and limitations of the technique.

Contributed Papers

2:40

4pAB5. An experimental evaluation of the performance of acoustic recording systems for estimating avian species richness and abundance. Antonio Celis Murillo (Natural Resources and Environmental Sci., Univ. of Illinois at Urbana-Champaign, 1704 Harrington Dr., Champaign, IL 61821, celismu1@illinois.edu), Jill Deppe (Biological Sci., Eastern Illinois Univ., Champaign, IL), Jason Riddle (Natural Resources, Univ. of Wisconsin at Stevens Point, Stevens Point, WI), Michael P. Ward (Natural Resources and Environmental Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Theodore Simons (USGS cooperative fish and Wildlife Res. unit, North Carolina State Univ., Raleigh, NC)

Comparisons between field observers and acoustic recording systems have shown great promise for sampling birds using acoustics methods. Comparisons provide information about the performance of recording systems and field observers but do not provide a robust validation of their true sampling performance—i.e., precision and accuracy relative to known population size and richness. We used a 35-speaker bird song simulation system to experimentally test the accuracy and precision of two stereo (Telinga and SS1) and one quadraphonic recording system (SRS) for estimating species richness, abundance, and total abundance (across all species) of vocalizing birds. We simulated 25 bird communities under natural field conditions by placing speakers in a wooded area at 4–119 m from the center of the survey at differing heights and orientations. We assigned recordings randomly to one of eight skilled observers. We found a significant difference among microphones in their ability to accurately estimate richness ($p = 0.0019$) and total bird abundance ($p = < 0.0001$). Our study demonstrates that acoustic recording systems can potentially estimate bird abundance and species richness accurately; however, their performance is likely to vary by its technical characteristics (recording pattern, microphone arrangement, etc.).

2:55–3:15 Break

3:15

4pAB6. Spatial variation of the underwater soundscape over coral reefs in the Northwestern Hawaiian Islands. Simon E. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., 7038 Old Brentford Rd., Alexandria, VA 22310, simon.freeman@gmail.com), Lauren A. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), Marc O. Lammers (Oceanwide Sci. Inst., Honolulu, HI), and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Coral reefs create a complex acoustic environment, dominated by sounds produced by benthic creatures such as crustaceans and echinoderms. While there is growing interest in the use of ambient underwater biological sound as a gauge of ecological state, extracting meaningful information from recordings is a challenging task. Single hydrophone (omnidirectional) recorders can provide summary time and frequency information, but as the spatial distribution of reef creatures is heterogeneous, the properties of reef sound arriving at the receiver vary with position and arrival angle. Consequently, the locations and acoustic characteristics of individual sound producers remain unknown. An L-shaped hydrophone array, providing direction-and-range sensing capability, can be used to reveal the spatial variability of reef sounds. Comparisons can then be made between sound sources and other spatially referenced information such as photographic data. During the summer of 2012, such an array was deployed near four different benthic ecosystems in the Northwestern Hawaiian Islands, ranging from high-latitude coral reefs to communities dominated by algal turf. Using conventional and adaptive acoustic focusing (equivalent to curved-wavefront beamforming), time-varying maps of sound production from benthic organisms were created. Comparisons with the distribution of nearby sea floor features, and the makeup of benthic communities, will be discussed.

3:30

4pAB7. Density estimates of odontocetes in an active military base using passive acoustic monitoring. Bethany L. Roberts (School of Biology, Univ. of St. Andrews, Sea Mammal Res. Unit, St. Andrews, Fife KY16 8LB, United Kingdom, blr2@st-andrews.ac.uk), Zach Swaim, and Andrew J. Read (Duke Marine Lab, Duke Univ., Beaufort, NC)

We deployed passive acoustic monitoring devices in Camp Lejeune, North Carolina, USA, to estimate density of odontocete populations. Four C-PODs (echolocation click detectors) were deployed in water depths ranging from 13 to 21 meters from 30 November 2012 to 13 November 2013. Two species of odontocetes are known to inhabit the survey area: bottlenose dolphins and Atlantic spotted dolphins. These methods incorporate (i) the rate at which the animals produce echolocation cues, (ii) the probability of detecting cues, and (iii) the false positive rate of detections. To determine the cue rate of bottlenose dolphins, we attached DTAGs to 14 bottlenose dolphins during 2012 and 2013 in Sarasota, Florida. To determine cue rate of spotted dolphins, we used six recordings of focal follows from 2001-2003 in an area adjacent to C-POD deployment locations. Echolocation playbacks to C-PODs were used to obtain false positive rate and detection radius of each C-POD. Furthermore, we obtained proportions of bottlenose and spotted dolphins in the survey area from concurrent line transect surveys. Preliminary results indicate that dolphins were detected on all four C-PODs during every month of the survey period. Future studies in areas where multiple species are present could potentially use methods described here.

3:45

4pAB8. Preliminary calculation of individual echolocation signal emission rate of Franciscana dolphins (*Pontoporia blainvillei*). Artur Andriolo (Zoology Dept., Federal Univ. of Juiz de Fora, Universidade Federal de Juiz de Fora, Rua José Lourenço Kelmer, s/n - Campus Universitário Bairro São Pedro, Juiz de Fora, Minas Gerais 36036-900, Brazil, artur.andriolo@ufjf.edu.br), Federico Sucunza (Ecology Graduate Program, Federal Univ. of Juiz de Fora, Juiz de Fora, Brazil), Alexandre N. Zerbini (Ecology, Instituto Aqualie, Juiz de Fora, Brazil), Daniel Danilewicz (Zoology Graduate Program, State Univ. of Santa Cruz, Ilhéus, Brazil), Marta J. Cremer (Biological Sci., Univ. of Joinville Region, Joinville, Brazil), and Annelise C. Holz (Graduate Program in Health and Environment, Univ. of Joinville Region, Joinville, Brazil)

Calculation of echolocation signals emission rate is necessary to estimate how many individuals are vocalizing, especially if passive acoustic density estimation methods are to be implemented. We calculated the individual emission rate of echolocation signals of franciscana dolphin. Fieldwork was between 22 and 31 January of 2014 at Babitonga Bay, Brazil. Acoustic data and group size were registered when animals were within visual range at maximum distance of 50 meters. We used a Cetacean Research^{TM} hydrophone. The sound was digitized by Analogic/Digital IOtech, stored as wav-files and analyzed with Raven software. A band limited energy detector was set to automatically extract echolocation signals. The emission rate was calculated dividing the clicks registered for each file by the file duration and by the number of individuals in the group. We analyzed 240 min of sound of 36 groups. A total of 29,164 clicks were detected. The median individual click rate was 0.290 clicks/s (10th = 0.036 and 90th = 1.166 percentiles). The result is a general approximation of the individual echolocation signal emission rate. Sound production rates are potentially dependent on a number of factors, like season, group size, sex, or even density itself. [This study was supported by IWC/Australia, Petrobras, Fundo de Apoio à Pesquisa/UNIVILLE.]

4:00

4pAB9. Investigating the potential of a wave glider for cetacean density estimation—A Scottish study. Danielle Harris (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, The Observatory, Buchanan Gardens, St. Andrews KY16 9LZ, United Kingdom, dh17@st-andrews.ac.uk) and Douglas Gillespie (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, United Kingdom)

A major advantage of autonomous vehicles is their ability to provide both spatial and temporal coverage of an area during a survey. However, there is a need to assess whether these technologies are suitable for monitoring cetacean population densities. Data are presented from a Wave Glider deployed off the east coast of Scotland between March and April 2014. Key areas of survey design, data collection, and analysis were investigated. First, the ability of the glider to complete a designed line transect survey was assessed. Second, the encounter rates of all detected species were estimated. Harbour porpoise (*Phocoena phocoena*) was the most commonly encountered species and became the focal species in this study. Using the harbor porpoise encounter rate, the amount of survey effort required to estimate density with a suitable level of uncertainty was estimated. A separate experiment was designed to estimate the average probability of harbor porpoise detection by the glider. The glider was deployed near an array of nine C-PODs (odontocete detection instruments) and the same harbor porpoise click events were matched across instruments. Such matches can be analyzed using spatially explicit capture recapture methods, which allow the detection efficiency of the glider to be estimated.

4:15

4pAB10. Toward acoustically derived population estimates in marine conservation: An application of the spatially-explicit capture-recapture methodology for North Atlantic right whales. Danielle Cholewiak, Steven Brady, Peter Corkeron, Genevieve Davis, and Sofie Van Parijs (Protected Species Branch, NOAA Northeast Fisheries Sci. Ctr., 166 Water St., Woods Hole, MA 02543, danielle.cholewiak@noaa.gov)

Passive acoustics provide a flexible tool for developing understanding of the ecology and behavior of vocalizing marine animals. Yet despite a robust capacity for detecting species presence, our ability to estimate population abundance from acoustics still remains poor. Critically, abundance estimates are precisely what conservation practitioners and policymakers often require. In the current study, we explored the application of acoustic data in the spatially-explicit capture-recapture (SECR) methodology, to evaluate whether acoustics can be used to infer abundance in the endangered North Atlantic right whale. We sub-sampled a year-long acoustic dataset from archival recorders deployed in Massachusetts Bay. Multichannel data were reviewed for the presence of up-calls. A total of 1659 unique up-calls were detected. Estimates of up-call density ranged from zero to 608 ($\pm$ 70 SE) up-calls/hour. Estimates of daily abundance, when corrected for average calling rate, ranged from 0—69 ($\pm$ 21 SE) individuals per day. These results qualitatively compare well with patterns in right whale occurrence reported from aerial-based visual surveys. Since acoustic abundance calculations are affected by variation in calling behavior, estimates should be interpreted cautiously; however, these results indicate that passive acoustics has the potential to directly inform conservation and management strategies.

4:30

4pAB11. Statistical mechanics techniques applied to the analysis of humpback whale inter-call intervals. Gerald L. D'Spain (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Tyler A. Helble (SPAWAR SSC Pacific, San Diego, CA), Heidi A. Batchelor, and Dennis Rimington (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

Techniques developed in statistical mechanics recently have been applied to the analysis of the topology of complex human communication networks. These methods examine the network's macroscopic statistical properties rather than the details of individual interactions. Here, these methods are applied to the analysis of the time intervals between humpback whale calls detected in passive acoustic monitoring data collected by the bottom-mounted hydrophones on the Pacific Missile Range Facility (PMRF) west of Kauai, Hawaii. Recently developed localization and tracking algorithms for use with PMRF data have been applied to separate the calls of an individual animal from those of a collection of animals. As with the distributions of time intervals between human communications, the distributions of time intervals between humpback whale call detections are distinctly different than those expected for a purely independent, random (Poisson) process. This conclusion holds both for time intervals between calls from individual animals and from the collection of animals vocalizing simultaneously. although significant differences in these probability distributions occur. A model based on the migration of clusters of animals is developed to fit the distributions. Possible mechanisms giving rise to aspects of the distributions are discussed. [Work supported by the Office of Naval Research, Code 322-MMB.]

4:45–5:15 Panel Discussion

Session 4pBA

Biomedical Acoustics: Mechanical Tissue Fractionation by Ultrasound: Methods, Tissue Effects, and Clinical Applications II

Vera A. Khokhlova, Cochair
University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Jeffrey B. Fowlkes, Cochair
Univ. of Michigan Health System, 3226C Medical Sciences Building I, 1301 Catherine Street, Ann Arbor, MI 48109-5667

Invited Papers

1:30

4pBA1. High intensity focused ultrasound-induced bubbles stimulate the release of nucleic acid cancer biomarkers. Tatiana Khokhlova (Medicine, Univ. of Washington, Harborview Medical Ctr., 325 9th Ave. Box 359634, Seattle, WA 98104, tdk7@uw.edu), John R. Chevillet (Inst. for Systems Biology, Seattle, WA), George R. Schade (Urology, Univ. of Washington, Seattle, WA), Maria D. Giraldez (Medicine, Univ. of Michigan, Ann Arbor, MI), Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Joo Ha Hwang (Medicine, Univ. of Washington, Seattle, WA), and Muneesh Tewari (Medicine, Univ. of Michigan, Ann Arbor, MI)

Recently, several nucleic acid cancer biomarkers, e.g., microRNA and mutant DNA, have been identified and shown promise for improving cancer diagnostics. However, the abundance of these biomarker classes in the circulation is low, impeding reliable detection and adoption into clinical practice. Here, the ability of HIFU-induced bubbles to stimulate release of cancer-associated microRNAs by tissue fractionation or permeabilization was investigated in a heterotopic syngeneic rat prostate cancer model. A 1.5 MHz HIFU transducer was used to either mechanically fractionate subcutaneous tumor with boiling histotripsy (BH) (~20 kW/cm^2, 10 ms pulses, and duty factor 0.01) or to permeabilize tumor tissue with inertial cavitation activity (p- = 16 MPa, 1 ms pulses, duty factor 0.001). Blood was collected immediately prior to and serially up to 24-hours after treatments. Plasma concentrations of microRNAs were measured by quantitative RT-PCR. Both exposures resulted in a rapid (within 15 min), short ($\leq$3 h) and dramatic (over ten-fold) increase in relative plasma concentrations of tumor-associated microRNAs, Histologic examination of excised tumor confirmed complete fractionation of targeted tumor by BH and localized areas of intraparenchymal hemorrhage and tissue disruption by cavitation-based treatment. These data suggest a clinically useful application of HIFU-induced bubbles for non-invasive molecular biopsy. [Grant support: NIH 1K01EB015745, R01CA154451, R01DK085714.]

1:50

4pBA2. Tissue decellularization with boiling histotripsy and the potential in regenerative medicine. Yak-Nam Wang (APL, CIMU, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ynwang@u.washington.edu), Tatiana Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Adam Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Wayne Kreider (APL, CIMU, Univ. of Washington, Seattle, WA), Ari Partanen (Clinical Sci. MR Therapy, Philips Healthcare, Andover, Maryland), Navid Farr (Dept. of BioEng., Univ. of Washington, Seattle, WA), George Schade (Dept. of Urology, Univ. of Washington, Seattle, WA), Michael Bailey (APL, CIMU, Univ. of Washington, Seattle, WA), and Vera Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

There have been major advances in the development of replacement organs by tissue engineering (TE); however, one of the holy grails is still in the development of biomimetic structures that replicate the complex 3-D vasculature. Creation of bioartificial organs by decellularization shows greater promise in reaching the clinic compared to TE. However, current decellularization techniques require the use of chemical and biological agents, often in combination with physical force, which could result in damage to the matrix. Here we evaluate the use of boiling histotripsy (BH) to selectively decellularize large volumes of tissue. BH lesions (10–20 mm diameter) were produced in bovine liver with a clinical 1.2 MHz MR-HIFU system (Sonalleve, Philips, Finland), using thirty 10 ms pulses, and pulse repetition frequencies of 1–10 Hz. Peak acoustic powers corresponding to an estimated *in situ* shock front amplitude of 65 MPa were used. Macroscopic and histological evaluation revealed treatment conditions that produced decellularized lesions in which major fibrous structures such as stroma and vasculature remained intact while parenchymal cells were mostly lyzed. With further tailoring of the pulsing scheme parameters, this treatment modality could potentially be optimized for organ decellularization. [Work supported by NIH EB007643, K01-EB-015745-01, T32-DK007779, and NSBRI NASA-NCC 9-58.]

2:10

4pBA3. Destruction of microorganisms by high-energy pulsed focused ultrasound. Timothy A. Bigelow (Elec. and Comput. Eng., Mech. Eng., Iowa State Univ., 2113 Coover Hall, Ames, IA 50011, bigelow@iastate.edu)

The use of high-energy ultrasound pulses to generate and excite clouds of microbubbles has shown great potential to mechanically destroy soft tissue in a wide range of clinical applications. In our work, we have focused on extending the application of cavitation based histotripsy to the destruction of microorganisms such as bacteria biofilms and microalgae. Bacteria biofilms pose a significant problem when treating infections on medical implants while the fractionation of microalgae in an efficient manner could lower the production cost of biofuels. In the past, we have shown a 4.4-log10 reduction of viable *Escherichia coli* bacteria capable of forming a colony in a biofilm following a high-energy pulsed focused ultrasound exposure. We have also shown complete removal of *Pseudomonas aeruginosa* biofilms from a Pyrolytic graphite substrate based on fluorescence imaging following live/dead staining. We also showed minimal temperature increase when the appropriate ultrasound pulse parameters were utilized. Recently, we have shown that high-energy pulsed ultrasound at 1.1 MHz can fractionate the microalgae model system *Chlamydomonas reinhardtii* for lipid extraction/biofuel production in both flow and stationary exposure systems with improved efficiency over traditional sonicators. In these studies, the fractionation of the cells was quantified by protein and chlorophyll release following exposure.

Contributed Papers

2:30

4pBA4. Dependence of ablative ability of high-intensity focused ultrasound cavitation-based histotripsy on mechanical properties of agar. Jin Xu (Eng., John Brown Univ., Siloam Springs, AR), Timothy Bigelow (Elec. and Comput. Eng., Iowa State Univ., Iowa State University, 2113 Coover Hall, Ames, IA 50011, bigelow@iastate.edu), Gabriel Davis, Alex Avendano, Pranav Shrotriya, Kevin Bergler (Mech. Eng., Iowa State Univ., Ames, IA), and Zhong Hu (Elec. and Comput. Eng., Iowa State Univ., Ames, IA)

Cavitation-based histotripsy uses high-intensity focused ultrasound (HIFU) at low duty factor to create bubble clouds inside tissue to liquefy a region and provides better fidelity to planned lesion coordinates and the ability to perform real-time monitoring. The goal of this study was to identify the most important mechanical properties for predicting lesion dimensions, among these three: Young's modulus, bending strength, and fracture toughness. Lesions were generated inside tissue-mimicking agar, and correlations were examined between the mechanical properties and the lesion dimensions, quantified by lesion volume and by the width and length of the equivalent bubble cluster. Histotripsy was applied to agar samples with varied properties. A cuboid of 4.5 mm width (lateral to focal plane) and 6 mm depth (along beam axis) was scanned in a raster pattern with respective step sizes of 0.75 mm and 3 mm. The exposure at each treatment location was 15 s, 30 s, or 60 s long. Results showed that only Young's modulus influenced histotripsy's ablative ability and was significantly correlated with lesion volume and bubble cluster dimensions. The other two properties had negligible effects on lesion formation. Also, exposure time differentially affected the width and depth of the bubble cluster volume.

2:45

4pBA5. Shear waves induced by Lorentz force in soft tissues. Stefan Catheline, Graland-Mongrain Pol, Ali Zorgani, Remi Souchon, Cyril Lafon, and Jean-yves Chapelon (LabTAU, INSERM, Univ. of Lyon, 151 cours albert thomas, Lyon 69003, France, stefan.catheline@inserm.fr)

This study presents the observation of elastic shear waves generated in soft solids using a dynamic electromagnetic field. The first and second experiments of this study show that Lorentz force can induce a displacement in a soft phantom and that this displacement is detectable by an ultrasound scanner using speckle-tracking algorithms. For a 100 mT magnetic field and a 10 ms, 100 mA peak-to-peak electrical burst, the displacement reached a magnitude of 1 m. In the third experiment, we show that Lorentz force can induce shear waves in a phantom. A physical model using electromagnetic and elasticity equations is proposed and computer simulations are in good agreement with experimental results. The shear waves induced by Lorentz force are used in the last experiment to estimate the elasticity of a swine liver sample.

3:00–3:15 Break

3:15

4pBA6. Acoustic field characterization of the Waterlase2: Acoustic characterization and high speed photomicrography of a clinical laser generated shock wave therapy device for the treatment of periodontal biofilms in orthodontics and periodontics. Camilo Perez, Yak-Nam Wang (BioEng. and Ctr. for Industrial and Medical Ultrasound, CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, camipiri@uw.edu), Alina Sivriver, Dmitri Boutoussov, Vladimir Netchitailo (Biolase Inc., Irvine, CA), and Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Recent applications in endodontics and periodontics use erbium solid state lasers with fiber delivery in order to effectively kill bacteria and biofilms. In this paper, the acoustic field together with the bubble dynamics of a clinical portable Er,Cr:YSGG laser-generating device (Waterlase 2) was characterized. Field mapping with a calibrated PVDF hydrophone together with high speed imaging were performed in water for two different tip geometries (flat or tapered), three different tip diameters (200, 300, or 400μm), and two different laser pulse durations (60 or 700μs) at several laser pulse energy settings (5 mJ–400 mJ) for individual pulses and at different pulse repetition frequencies (5, 20, and 100 Hz). Peak positive pressures 5–50 mm away from the tip ranged from 0.1 to 2MPa, while peak negative pressures ranged from 0.1 to 1.2 MPa. There was a strong correlation between the acoustic emissions generated by the bubble and the high speed imaging dynamics of the bubble. An initial thermoelastic response, initial bubble collapse and further rebounds where analyzed individually and compared across different test parameters. For the initial thermoelastic pulse (laser generated), pulse rise times ranged from 40 to 200 ns. Differences between flat and tapered tips will be discussed.

3:30

4pBA7. Simulations of focused shear shock waves in soft solids and the brain. Bruno Giammarinaro, François Coulouvrat, and Gianmarco Pinton (Institut Jean le Rond d'Alembert UMR 7190, CNRS, Université Pierre et Marie Curie, d'alembert, case 162, 4, Pl. Jussieu, Paris cedex 05 75252, France, bruno.giam@hotmail.fr)

Because of a very small speed, shear waves in soft solids are extremely nonlinear, with nonlinearities four orders of magnitude larger than in classical solids. Consequently, these nonlinear shear waves can transition from a smooth to a shock profile in less than one wavelength. We hypothesize that traumatic brain injuries (TBI) could be caused by the sharp gradients resulting from shear shock waves. However, shear shock waves are not currently modeled by simulations of TBI. The objective of this paper is to describe shear shock wave propagation in soft solids within the brain, with source geometry determined by the skull. A 2D nonlinear paraxial equation with cubic nonlinearities is used as a starting point. We present a numerical scheme based on a second order operator splitting which allows the application of optimized numerical methods for each terms. We then validate the scheme with Guiraud's nonlinear self-similarity law applied to cusped caustics. Once validated, the numerical scheme is then applied to a blast wave

problem. A CT measurement of the human skull is used to determine the initial conditions and shear shock wave simulations are presented to demonstrate the focusing effects of the skull geometry.

3:45

4pBA8. Tissue damage produced by cavitation: The role of viscoelasticity. Eric Johnsen (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48104, ejohnsen@umich.edu) and Matthew Warnez (Eng. Phys., Univ. of Michigan, Ann Arbor, MI)

Cavitation may cause damage at the cellular level in a variety of medical applications, e.g., therapeutic and diagnostic ultrasound. While cavitation damage to bodies in water has been studied for over a century, the dynamics of bubbles in soft tissue remain vastly unexplored. One difficulty lies in the viscoelasticity of tissue, which introduces additional physics and time scales. We developed a numerical model to investigate acoustic cavitation in soft tissue, which accounts for liquid compressibility, full thermal effects, and viscoelasticity (including nonlinear relaxation and elasticity). The bubble dynamics are represented by a Keller-Miksis formulation and a spectral collocation method is used to solve for the stresses in the surrounding medium. Our numerical studies of a gas bubble exposed to a relevant waveform indicate that under inertial conditions high pressures and velocities are generated at collapse, though they are lower than those observed in water due to the elasticity and viscosity of the medium. We further find that significant deviatoric stresses and increased heating in tissue are attributable to viscoelasticity, due to material properties and different bubble responses compared to water.

4:00

4pBA9. Comparison of Gilmore-Akulichev's, Keller-Miksis's and Rayleigh-Plesset's equations on therapeutic ultrasound bubble cavitation. Zhong Hu (Elec. and Comput. Eng., Mech. Eng., Iowa State Univ., 2201 Coover Hall, Ames, IA 50011, zhonghu@iastate.edu), Jin Xu (Eng., John Brown Univ., Siloam Springs, AR), and Timothy A. Bigelow (Elec. and Comput. Eng., Mech. Eng., Iowa State Univ., Ames, IA)

Many models have been utilized to simulate inertial cavitation for ultrasound therapies such as histotripsy. The models range from the very simple Rayleigh-Plesset model to the complex Gilmore-Akulichev model. The computational time increases with the complexity of the model, so it is important to know when the results from the simpler models are sufficient. In this paper the simulation performance of the widely used Rayleigh-Plesset model, Keller-Miksis model, and Gilmore-Akulichev model both with and without gas diffusion are compared by calculating the bubble radius response and bubble wall velocity as a function the ultrasonic pressure and frequency. The bubble oscillates similarly with the three models within the first collapse for small pressures (<3MPa), but the Keller-Miksis model diverges at higher pressures. In contrast, the maximum expansion radius of the bubble is similar at all pressures with Rayleigh-Plesset and Gilmore-AKulichev although the collapse velocity is unrealistically high with Rayleigh-Plesset model. After multiple cycles, the Rayleigh-Plesset model starts to behave disparately both in the expansion and collapse stages. The inclusion of rectified gas diffusion lengthens the collapse time and increases the expansion radius. However, for frequency smaller than 1 MHz, the impact of gas diffusion is not significant.

4:15

4pBA10. Removal of residual bubble nuclei to enhance histotripsy soft tissue fractionation at high rate. Alexander P. Duryea, Charles A. Cain (Biomedical Eng., Univ. of Michigan, 2131 Gerstacker Bldg., 2200 Bonisteel Blvd., Ann Arbor, MI 48109, duryalex@umich.edu), William W. Roberts (Urology, Univ. of Michigan, Ann Arbor, MI), and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Previous work has shown that the efficacy of histotripsy soft tissue fractionation is dependent on pulse repetition frequency, with histotripsy delivered at low rates producing more efficient homogenization of the target volume in comparison to histotripsy delivered at high rates. This is attributed to the cavitation memory effect: microscopic residual cavitation nuclei that persist for hundreds of milliseconds following bubble cloud collapse can seed the repetitive nucleation of cavitation at a discrete set of sites within the target volume, producing heterogeneous lesion development. To mitigate this effect, we have developed low amplitude (MI<1) acoustic pulses to actively remove residual nuclei from the field. These bubble removal pulses utilize the Bjerknes forces to stimulate the aggregation and subsequent coalescence of remnant nuclei, consolidating the population from a very large number to a countably small number of remnant bubbles within several milliseconds. The effect is attainable in soft tissue mimicking phantoms following a very minimal degree of fractionation (within the first ten histotripsy pulses). Incorporation of this bubble removal scheme in histotripsy tissue phantom treatments at high rate (100 pulses/second) resulted in highly homogeneous lesions that closely approximated those achieved using an equal number of pulses applied at low rate (1 pulse/second); lesions generated at high rate without bubble removal had heterogeneous structure with increased collateral damage.

4:30

4pBA11. Two-dimensional speckle tracking using zero phase crossing with Riesz transform. Mohamed Khaled Almekkawy (Elec. Eng., Western New England, 2056 Knapp St., Saint Paul, MN 55108, alme0078@umn.edu), Yasaman Adibi, Fei Zheng (Elec. Eng., Univ. of Minnesota, Minneapolis, MN), Mohan Chirala (Samsung Res. America, Richardson, TX), and Emad S. Ebbini (Elec. Eng., Univ. of Minnesota, Minneapolis, MN)

Ultrasound speckle tracking provides robust estimates of fine tissue displacements along the beam direction due to the analytic nature of echo data. We introduce a new multi-dimensional ST method (MDST) with subsample accuracy in all dimensions. The algorithm based on the gradient of the magnitude and the zero-phase crossing of 2D complex correlation of the generalized analytic signal. The generalization method utilizes the Riesz transform which is the vector extension of the Hilbert transform. Robustness of the tracking algorithm is investigated using a realistic synthetic data sequences created with (Field II) for which the bench mark displacement was known. In addition, the new MDST method is used in the estimation of the flow and surrounding tissue motion on human carotid artery *in vivo*. The data was collected using a linear array probe of a Sonix RP ultrasound scanner at 325 fps. The vessel diameter has been calculated from the upper and lower vessel walls displacement, and clearly shows a blood pressure wave like pattern. The results obtained show that using Riesz transform produces more robust estimation of the true displacement of the simulated model compared to previously published results. This could have significant impact on strain calculations near vessel walls.

4:45

4pBA12. 1-MHz ultrasound stimulates *in vitro* production of cardiac and cerebrovascular endothelial cell vasodilators. Azzdine Y. Ammi (Knight Carviovascular Inst., OHSU, 3181 SW Sam Jackson Park Rd., Portland, OR 97239, ammia@ohsu.edu), Catherine M. Davis (Dept. of Anesthesiology and Perioperative Medicine, OHSU, Portland, OR), Brian Mott (Knight Carviovascular Inst., OHSU, Portland, OR), Nabil J. Alkayed (Dept. of Anesthesiology and Perioperative Medicine, OHSU, Portland, OR), and Sanjiv Kaul (Knight Carviovascular Inst., OHSU, Portland, OR)

Ultrasound exposure of the heart and brain during vessel occlusion reduces infarct size. Our aim was to study the production of vasodilatory compounds by endothelial cells after ultrasound stimulation. A 1.05-MHz single element transducer was used to insonify primary mouse endothelial cells (ECs) from heart and brain with a 50-cycle tone burst at a pulse repetition frequency of 50 Hz. Two time points were studied after ultrasound exposure: 15 and 45 minutes. In heart ECs, EETs levels increased significantly with 0.5 MPa (139 $\pm$ 16%, $p<0.05$) and 0.3 MPa (137 $\pm$ 15%, $p<0.05$) at 15 and 45 min post stimulation, respectively. HETEs and DHETs did not change significantly. There was a trend toward increased adenosine, with maximum release at 0.5 MPa (332 $\pm$ 73% vs. 100% control, $p<0.05$). The trend toward increased eNOS phosphorylation was greater at 15 than 45 min. In brain ECs adenosine release was increased, however increased eNOS phosphorylation was not significant. 11, 12- and 14, 15- EETs were increased while 5- and 15-HETEs were decreased. Pulsed ultrasound at 1.05 MHz has the ability to increase adenosine, p-eNOS, and EET production by cardiac and cerebrovascular ECs. Interestingly, in brain ECs, the vasoconstricting HETEs were decreased.

5:00

4pBA13. Ultrasound-induced fractionation of the intervertebral disk. Delphine Elbes, Olga Boubriak, Shan Qiao, Michael Molinari (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Jocelyn Urban (Dept. of Physiol., Anatomy and Genetics, Univ. of Oxford, Oxford, United Kingdom), Robin Cleveland, and Constantin Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Oxford, Oxfordshire, United Kingdom, constantin.coussios@eng.ox.ac.uk)

Current surgical treatments for lower back pain, which is strongly associated with degeneration of the intervertebral disk, are highly invasive and have low long-term success rates. The present work thus aims to develop a novel, minimally invasive therapy for disk replacement without the need for surgical incision. Using *ex vivo* bovine coccygeal spinal segments as an experimental model, two confocally aligned 0.5 MHz HIFU transducers were positioned with their focus inside the disc and used to generate peak rarefactional pressures in the range of 1–12 MPa. Cavitation activity was monitored, characterized, and localized in real time using both a single-element passive cavitation detector and a 2D Passive Acoustic Mapping array. The inertial cavitation threshold in the central portion of the disk, the nucleus pulposus (NP), was first determined both in the absence and in the presence of externally injected cavitation nuclei. HIFU exposure parameters were subsequently optimized to maximize sustained inertial cavitation over 10 min and achieve fractionation of the NP. Following sectioning of treated disks, staining of live and dead cells as well as microscopy under polarized light were used to assess the impact of the treatment on cell viability and collagen structure within the NP, inner annulus and outer annulus.

THURSDAY AFTERNOON, 30 OCTOBER 2014 — MARRIOTT 9/10, 1:30 P.M. TO 4:00 P.M.

Session 4pEA

Engineering Acoustics: Acoustic Transduction: Theory and Practice II

Roger T. Richards, Chair
US Navy, 169 Payer Ln, Mystic, CT 06355

Contributed Papers

1:30

4pEA1. Vibration sensitivity measurements of silicon and acoustic-gradient microphones. Marc C. Reese (Harman Embedded Audio, Harman Int.., 6602 E 75th St. Ste. 520, Indianapolis, IN 46250, marc.reese@harman.com)

Microphones are often required to record audio while in a vibration environment. Therefore, it is important to maximize the acoustic-to-vibration sensitivity of such microphones. It has previously been shown that the vibration sensitivity of a microphone is, to first order, proportional to the mass per unit area of the diaphragm including the air loading effect. Although the air loading is generally minimal for omnidirectional condenser microphones with thick diaphragms, these measurements show that it cannot be ignored for newer silicon-based micro-electro-mechanical-system (MEMS) and acoustic-gradient microphones. Additionally, since microphone vibration sensitivities are typically not reported by microphone manufacturers, nor measured using standardized equipment, the setup of an inexpensive vibration measurement apparatus and associated challenges are discussed.

1:45

4pEA2. Non-reciprocal acoustic devices based on spatio-temporal angular-momentum modulation. Romain Fleury, Dimitrios Sounas, and Andrea Alu (ECE Dept., The Univ. of Texas at Austin, 1 University Station C0803, Austin, TX 78712, romain.fleury@utexas.edu)

Acoustic devices that break reciprocity, for instance acoustic isolators or circulators, may find exciting applications in a variety of fields, including imaging, acoustic communication systems, and noise control. Non-reciprocal acoustic propagation has typically been achieved using non-linear phenomena, which require high input power levels and introduce distorsions. In contrast, we have recently demonstrated compact linear isolation for audible airborne sound by means of angular momentum bias [Fleury *et al*., Science 343, 516 (2014)], exploiting modal splitting in a ring cavity polarized by an internal, constantly circulating fluid, whose motion is imparted using low-noise CPU fans. We present here an improved design with no moving parts, which is directly scalable to ultrasonic frequencies and fully integrable. Instead of imparting angular momentum in the form of a moving medium as in our previous approach, we make use of spatio-temporal acoustic modulation of three coupled acoustic cavities, a strategy that can be readily implemented in integrated ultrasonic devices, for instance, using piezoelectric effects. In this new paradigm, the required modulation frequency is orders of magnitude lower than the signal frequency, and the modulation efficiency is maximized. This constitutes a pivotal step towards practically realizing compact, linear, noise-free, tunable non-reciprocal acoustic components for full-duplex acoustic communications and isolation.

2:00

4pEA3. An analysis of multi-year acoustic and energy performance data for bathroom and utility residential ventilation fans. Wongyu Choi, Antonio Gomez, Michael B. Pate, and James F. Sweeney (Mech. Eng., Texas A&M Univ., 2401 Welsh Ave. Apt. 615, 615, College Station, TX 77845, wongyuchoi@tamu.edu)

Loudness levels have been established as a new requirement in residential ventilation standards and codes including ASHRAE and IECC. Despite the extensive application of various standards and codes, the control of loudness has not been a common target in past whole-house ventilation standards and codes. In order to evaluate the appropriate loudness of ventilation fans, especially in terms of leading standards and codes, a statistical analysis is necessary. Therefore, this paper provides statistical data for bathroom and utility ventilation fans over a nine year period from 2005 to 2013. Specifically, this paper presents an evaluation of changes in fan loudness over the 9 year test period and the relevance of loudness to leading standards including HVI and ASHRAE. The loudness levels of brushless DC-motor fans are also evaluated in comparison to the loudness of AC-motor fans. For AC and DC motor fans, relationships between loudness and efficacy was determined and then explained with regression models. Based on observations, this paper introduces a new "loudness-to-energy ratio" coefficient, L/E, which is a measure of the acoustic and energy performance of a fan. Relationships between acoustic and energy performances are established by using L/E coefficients with supporting statistics for bathroom and utility fans.

2:15

4pEA4. Non contact ultrasound stethoscope. Nathan Jeger, Mathias Fink, and Ros Kiri Ing (Institut Langevin, ESPCI ParisTech, 1 rue Jussieu, Paris 75005, France, nathan.jeger@espci.fr)

Heartbeat and respiration are very important vital signs that indicate health and psychological states of a person. Recent technologies allow to detect both physical parameters on a human subject by using different techniques with and without contact. Noncontact systems often use electromagnetic waves for contactless measurement but approaches based on ultrasound waves, laser or video processes are also proposed. In this abstract an alternative ultrasound system for non-contact and local measurement is presented. The system works in echographic mode and ultrasound signals are processed using two methods. The experimental setup uses an elliptic mirror to focus ultrasonic waves onto the skin surface. Backscattered waves are recorded by a microphone located close to the emitting transducer. Heartbeat and respiration signals are determined from the skin displacement caused by the chest-wall motion. For comparison purpose, the cross-correlation method, which uses broadband signal, and the Doppler method, which uses narrowband signal, are applied to measure the skin displacement. Sensitivity and accuracy parameters of the two methods are compared. At least, as the measurement is local, the system can act as a noncontact stethoscope to listen the internal sounds of the human body even through the light clothes of the patient.

2:30

4pEA5. High sensitivity imaging of resin-rich regions in graphite/epoxy laminates using joint entropy. Michael Hughes (Int. Med./Cardiology, Washington Univ. School of Medicine, School of Medicine Campus Box 8215, St. Louis, MO 63108, mshatctrain@gmail.com), John McCarthy (Mathematics, Washington Univ., St. Louis, MO), Jon Marsh, and Samuel Wickline (Int. Med./Cardiology, Washington Univ. School of Medicine, Saint Louis, MO)

The continuing difficulty of detecting critical flaws in advanced materials requires novel approaches that enhance sensitivity to defects that might impact performance. This study compares different approaches for imaging a near-surface resin-rich defect in a thin graphite/epoxy plate using backscattered ultrasound. The specimen, having a resin-rich void immediately below the top surface ply, was scanned with a 1 in. dia., 5 MHz center frequency, and 4 in. focal length transducer. A computer controlled apparatus comprised of an x-y-z motion controller, a digitizer (LeCroy 9400A), and an ultrasonic pulser/receiver (Panametrics 5800) was used to acquire data on a 100 $\times$100 grid of points covering a 3 $\times$ 3 in. square. At each grid point 256 512-word, 8-bit backscattered waveforms, were digitized, signal averaged, and then stored on computer for off-line analysis. The same backscattered waveforms were used to produce peak-to-peak, signal energy, as well as entropy images. All of the entropy images exhibit better border delineation and defect contrast than the either peak-to-peak or signal energy. The best results are obtained using the joint entropy of the backscattered waveforms with a reference function. Two different references are examined: a reflection from a stainless steel reflector, and an approximate optimum obtained from an iterative parametric search. The joint entropy images produced using the optimum reference exhibit ~3 times the contrast obtained in previous studies.

2:45

4pEA6. New compensation factors for the apparent propagation speed In transmission line matrix uniform grid meshes. Alexandre Brandao (Graduate Program in Elec. and Telecommunications Eng., Universidade Federal Fluminense, Rua Passo da Patria, 156, Sao Domingos, Niteroi, RJ 24210-240, Brazil, abrand@operamail.com), Edson Cataldo (Appl. Mathematics Dept., Universidade Federal Fluminense, Niteroi, RJ, Brazil), and Fabiana R. Leta (Mech. Eng. Dept., Universidade Federal Fluminense, Niteroi, RJ, Brazil)

Numerical models consisting of two-dimensional (2D) and three-dimensional (3D) uniform grid meshes for the Transmission Line Matrix Method (TLM), use sqrt(2) and sqrt(3), respectively, to compensate for the apparent sound speed. In this work, new compensation factors are determined from a priori simulations, performed without compensation, in 2D and 3D TLM one-section cylindrical waveguide acoustic models. The mistuned resonance peaks obtained from these simulations are substituted in the corresponding equations for the resonance frequencies in one-section cylindrical acoustical waveguides to find the mesh apparent sound speed and, thus, the necessary compensation. The TLM meshes are constructed over the voxels (Volumetric Picture Elements) of segmented MRI volumes, so that the extracted mesh fits the segmented object. The TLM method provides a direct simulation approach instead of solving a PDE by variational methods that must consider the plane wave assumption to run properly. Results confirm the improvement over the conventional compensation factors, particularly for frequencies above 4 kHz, providing a concrete reduction of the topology-dependent numerical dispersion for both 2D and 3D TLM lattices. Since this dispersion problem is common to all TLM applications using uniform grids, investigators in other areas of wave propagation can also benefit from these findings.

3:00–3:15 Break

3:15

4pEA7. A low-cost alternative power supply for integrated electronic piezoelectric transducers. Ricardo Brum, Sergio L. Aguirre, Stephan Paul, and Fernando Corrêa (Centro de Tecnologia, Universidade Federal de Santa Maria, Rua Erly de Almeida Lima, 650, Santa Maria, RS 97105-120, Brazil, ricardozbrum@yahoo.com.br)

Comercial hardware compatible with IEPE precision sensors normally are expensive and often coupled to proprietary and expensive software packages. commercially available sound cards are a low cost option for AD, but are incompatible with IEPE sensors. To create 4 mA constant current for IEPE transducers commercial solutions are available and labs also have created such solutions, e.g., ITA at RWTH Aachen University. Unfortunately, commercially available circuits are still to expensive for large scale classroom use in Brazil and circuits created elsewhere contain parts subject to US export restrictions or require machines for creation of circuits. Thus, based on a previous project, a new low-cost prototype was mounted on phenolic board. The circuit was tested with an IEPE microphone connected to a commercial soundcard and ITA-Toolbox software and compared to a commercial hardware/software package. The results were very similar in the frequency range between 20 Hz and 10 kHz. The difference below 20 Hz probably occurs due the different high pass filters in the AD-cards. The differences in the high frequency range are very likely due to differences in the electrical background noise. The results suggest the device works well and is a good alternative to make measurements with IEPE sensors.

3:30

4pEA8. Determination of the characteristic impedance and the complex wavenumber of an absorptive material used in dissipative silencer. Key F. Lima, Nilson Barbieri (Mech. Eng., PUCPR, Imaculada Conceição, 1155, Curitiba, Paraná 80215901, Brazil, keyflima@gmail.com), and Renato Barbieri (Mech. Eng., UDESC, Joinville, Brazil)

The silencers are acoustic filters that have the purpose of reducing unwanted noise emitted by engines or equipment to acceptable levels. The vehicular silencers have large volume and dissipative properties. Dissipative silencers have absorptive material inside. These materials are typically fibrous and have good acoustic dissipation. However, few works depict the acoustic behavior of silencers with absorptive materials. The difficulty in evaluating this type of silencer is determining the acoustic properties of the absorptive material: the characteristic impedance and the complex wavenumber. This work shows an inverse methodology for determining the acoustic properties of the absorptive material used in silencers. First, it is found the silencer's acoustic efficiency in terms of the experimental sound transmission loss. Second, the absorptive material properties are determined with a parameters adjustment through a direct search optimization algorithm. In this step, the adjustment is done by applying The Finite Element Method in the search for the silencer's computational efficiency. The final step is to verify the difference between the experimental and computational results. For this work is used the acoustic efficiency of a silencer that has already been published in the literature. The results show good agreement.

3:45

4pEA9. Flat, lightweight, transparent acoustic transducers based on dielectric elastomer and gel. Kun Jia (The State Key Lab. for Strength and Vib. of Mech. Structures, Xian Jiaotong Univ., South 307 Rm., 1st Teaching Bldg.,West of the Xianning Rd. No.28, Xian, Shannxi 710049, China, kunjia@mail.xjtu.edu.cn)

The advances in flat-panel displays and Super Hi-Vision with a 22.2 multichannel sound system exhibit an entirely new viewing and listening environment for the audience; however, flat and lightweight acoustic transducers are required to fulfill this prospect. In this paper, a flat lightweight acoustic transducer with a rather simple structure is proposed. Polyacrylic elastomer membrane (VHB4905, 3M corporation) with 4 mm diameter, 0.5 mm thickness is biaxially prestretched and fixed on a polyurethane ring as the vibrator, then ionic gel is painted on the center region of the membrane as electrodes, finally, conducting wires which are also made by ionic gel is attached to the edge of the electrodes for applying the AC voltage with a DC bias. The ultrahigh transmittance of the VHB4905, gel, and polyurethane makes the transducer totally transparent, which is of great interest in advanced media technology. The dynamic properties of the membrane are studied experimentally along with its acoustic performance. It has been found that the behavior of the dielectric elastomer membrane is quite complicated, both of the in plane and out of plane vibration mode exist. The transducer shows better performance below 10 kHz for the low elastic modular of the membrane.

THURSDAY AFTERNOON, 30 OCTOBER 2014 SANTA FE, 1:00 P.M. TO 4:10 P.M.

Session 4pMU

Musical Acoustics: Assessing the Quality of Musical Instruments

Andrew C. H. Morrison, Chair
Joliet Junior College, 1215 Houbolt Rd., Natural Science Department, Joliet, IL 60431

Invited Papers

1:00

4pMU1. Bamboo musical instruments: Some physical and mechanical properties related to quality. James P. Cottingham (Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

Bamboo is one of the most widely used materials in musical instruments, including string instruments and percussion as well as wind instruments. Bamboo pipe walls are complex, composed of a layered structure of fibers. The pipe walls exhibit non-uniformity in radial structure and density, and there is a significant difference between the elastic moduli parallel to and perpendicular to the bamboo fibers. This paper presents a summary of results from the literature on bamboo as a material for musical instruments. In addition, results are presented from recent measurements of the physical and mechanical properties of materials used in some typical instruments. In particular, a case study will be presented comparing measurements made on reeds and pipes from two Southeast Asian khaen. Of the two khaen discussed, one is a high quality khaen made by craftsmen in northeastern Thailand, while the other is an inexpensive instrument purchased at an import shop. For this pair of instruments, analysis and comparison have been made of the material properties of the bamboo pipes and the composition and mechanical properties of the metal alloy reeds.

1:20

4pMU2. Descriptive maps to illustrate the quality of a clarinet. Whitney L. Coyle (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu), Philippe Guillemain, Jean-Baptiste Doc, Alexis Guilloteau, and Christophe Vergez (Laboratroire de mécanique et d'acoustique, Marseille, France)

Generally, subjective opinions and decisions are made when judging the quality of musical instruments. In an attempt to become more objective, this research presents methods to numerically and experimentally create maps, over a range of control parameters, that describe instrument behavior for a variety of different sounds features or "quality markers" (playing regime, intonation, loudness, etc.). The behavior of instruments is highly dependent on the control parameters that are adjusted by the musician. Observing this behavior as a function of one control parameter (e.g., blowing pressure) can hide diversity of the overall behavior. An isovalue quality marker can be obtained for a multitude of control parameter combinations. Using multidimensional maps, where quality markers are a function of two or more control parameters, can solve this problem. Numerically: in two dimensions, a regular discretization on a subspace of control parameters can be implemented while conserving a reasonable calculation time. However, in higher dimensions (if, for example, aside from the blowing pressure and the lip force, we vary the reed parameters), it is necessary to use auto-adaptive sampling methods. Experimentally: the use of an artificial mouth allows us to maintain control conditions while creating these maps. We can also use an instrumented mouthpiece: this allows us to measure simultaneously and instantly these control parameters and create the maps "on the fly."

1:40

4pMU3. Recent works on the (psycho-)acoustics of wind instruments. Adrien Mamou-Mani (IRCAM, 1 Pl. Stravinsky, Paris 75004, France, adrien.mamou-mani@ircam.fr)

Two experiments aiming at linking acoustical properties and perception of wind instruments will be presented. The first one is a comparison between five oboes of the same model type. An original methodology is proposed, based on discrimination tests in playing conditions and default detection using acoustical measurements. The second experiment has been done on a simplified bass clarinet with an embedded active control system. A comparison of perceptual attributes, like sound color and playability, for different acoustical configurations (frequency and damping of resonances) is possible to test using a single system. A specific methodology and first results will be presented.

2:00

4pMU4. The importance of structural vibrations in brass instruments. Thomas R. Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu) and Wilfried Kausel (Inst. of Musical Acoust., Univ. of Music and Performing Arts, Vienna, Austria)

It is often thought that the input impedance uniquely determines the quality of a brass wind instrument. However, it is known that structural vibrations can also affect the playability and perceived sound produced by these instruments. The processes by which the structural vibrations affect the quality of brass instruments are not completely understood, but it is likely that vibrations of the metal couple to the lips as well as introducing small changes in the input impedance. We discuss the mechanisms by which structural vibrations can affect the quality of a brass instrument and suggest methods of incorporating these effects into an objective assessment of instrument quality.

2:20–2:40 Break

Contributed Papers

2:40

4pMU5. Investigating the colloquial description of sound by musicians and non-musicians. Jack Dostal (Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

What is meant by the words used in a subjective judgment of sound? Interpreting these words accurately allows these musical descriptions of sound to be related to scientific descriptions of sound. But do musicians, scientists, instrument makers, and others mean the same things by the same words? When these groups converse about qualities of sound, they often use an expansive lexicon of terms (bright, brassy, dark, pointed, muddy, etc.). It may be inaccurate to assume that the same terms and phrases have the same meaning to these different groups of people or even remain self-consistent for a single individual. To investigate the use of words and phrases in this lexicon, subjects with varying musical and scientific backgrounds were surveyed. The subjects were asked to listen to different pieces of recorded music and asked to use their own colloquial language to describe the musical qualities and differences they perceived in these pieces. In this talk, I describe some qualitative results of this survey and identify some of the more problematic terms used by these various groups to describe sound quality.

2:55

4pMU6. Chaotic behavior of the piccolo? Nicholas Giordano (Phys., Auburn Univ., College of Sci. and Mathematics, Auburn, AL 36849, njg0003@auburn.edu)

A direct numerical solution of the Navier-Stokes equations has been used to calculate the sound produced by a model of the piccolo. At low to moderate blowing speeds and at appropriate blowing angles, the sound pressure is approximately periodic with the expected frequency. As the blowing speed is increased or as the blowing angle is varied, the time dependence of the sound pressure becomes more complicated, and examination of the spectrum and the sensitivity of the sound pressure to initial conditions suggest that the behavior becomes chaotic. Similarities with the behavior found in Taylor-Couette and Rayleigh-Bènard instabilities of fluids are noted and possible implications for the nature of the piccolo tone are discussed.

3:10

4pMU7. Modeling 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 was 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 the individual and combined oscillators were then predicted and compared to measurements. The model predicts the mode shapes, natural frequencies, and damping well thus validating the modeling procedure. The effect of changing the cavity volume was then simulated to predict the behavior for a deeper air cavity.

3:25

4pMU8. Experiment to evaluate musical qualities of violin strings. Maxime Baelde (Acoust. & Environ. HydroAcoust. Lab, Université Libre de Bruxelles, 109 rue Barthélemy Delespaul, Lille 59000, France, maxime.baelde@centraliens-lille.org), Jessica De Saedeleer (Jessica De Saedeleer Luthier, Brussels, Belgium), and Jean-Pierre Hermand (Acoust. & Environ. hydroAcoust. lab, Université Libre de Bruxelles, Brussels, Belgium)

Most of violin strings on the market are made of different materials and size. They have different musical qualities: full, mellow, warm, and round, for example. Nevertheless, this description is subjective and related to string manufacturers. The aim of this study is to provide an experiment which gives an evaluation of the musical qualities of strings. This study is based on "musical descriptors," which gives information about a musical sound and psychoacoustics in order to match the musician point of view. "Musical descriptors" are also used for music classification. We use two sets of top-end strings model from two different brands. These strings are mounted on two similar violins and the strings are excited on their normal modes with harpsichord damper mechanism like and other means. The sound radiated is

recorded with a microphone and the vibration of the string with a coil-magnet device so as to have intrinsic and extrinsic string properties. Some musicians tried these strings and expressed what they thought about it. These acoustical and psychoacoustical analyzes will give information to the luthiers to know what string property allow one adjustment, in order to provide better advice aside from string manufacturers descriptions.

3:40

4pMU9. Vibration study of Indian folk instrument sambal. Ratnaprabha F. Surve (Phys., Nowrosjee Wadia College, 15 Tulip, Neco Gardens, Viman Nagar, Pune 21 411001, India, rfsurve@hotmail.com), Keith Desa (Phys., Nowrosjee Wadia College, 27, Maharashtra, India), and Dilip S. Joaj (Phys., Univ. of Pune, Pune, Maharashtra, India)

The percussion instruments family, in its folk category has many instruments like Dholki, Dimdi, Duff, Halagi, and Sambal. The Sambal is a folk membranophone made up of wood, played mainly in western India. Sambal a traditional drum, which is used in some religious functions. It is played by the people who are believed to be servants of goddess Mahalaxmi Devi. This instrument is made up of two approximately cylindrical wooden drums united along a common edge, having skin membranes stretched over their mouths. This instrument is played using two wooden sticks, of which one has a curved end. The right hand side drum's pitch is higher than the left. Its membrane is excited by striking repeatedly to generate sound of a constant pitch. This paper relates to vibrational analysis of the Sambal. A study has been carried out to check it's vibrational properties like modes of the vibration. The study is done by spectrum analysis (Fast Fourier Transform) using a simple Digital Storage Oscilloscope. The tonal quality of wood used for the cylinders and membrane is compared.

3:55

4pMU10. Experimental investigation of crash cymbal acoustic quality. Devyn P. Curley (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155), Zachary A. Hanan (Elec. Eng., Univ. of Colorado, Boulder, CO), Dan Luo (Mech. Eng., Tufts Univ., Medford, MA), Christopher W. Penny (Phys., Tufts Univ., Medford, MA), Christopher F. Rodriguez (Elec. and Comput. Eng., Tufts Univ., Medford, MA), Paul D. Lehrman (Music, Tufts Univ., Medford, MA), Chris B. Rogers, and Robert D. White (Mech. Eng., Tufts Univ., Medford, MA, r.white@tufts.edu)

A methodology to quantitatively evaluate the quality of the transmitted acoustic signature of cymbals is under development. High speed video recordings of a percussionist striking both a Zildjian 14 in. A-custom crash cymbal and a Zildjian Gen 16 low volume 16 in. crash cymbal were recorded and used to determine biometrically accurate crash and ride striking motions. A two degree of freedom robotic arm has been developed to mimic human striking motion. The robotic arm includes a high torque elbow joint driven in closed loop trajectory tracking and an impedance controlled wrist joint to approximate the variable stiffness of the stick grip. A quantitative comparison of robotic and human strikes will be made using high speed video. Repeatable strikes will be carried out using the robotic system in an anechoic chamber for different grades of Zildjian cymbals, including low volume Gen 16 cymbals. Acoustic features of the measured sound output will be compared to seek quantitative metrics for evaluating cymbal sound quality that compare favorably with the results of qualitative human assessments that are currently in use by the industry. Preliminary results indicate noticeable differences in cymbal acoustic output including variations in modal density, decay time, and beating phenomena.

THURSDAY AFTERNOON, 30 OCTOBER 2014 MARRIOTT 3/4, 1:15 P.M. TO 4:20 P.M.

Session 4pNS

Noise: Virtual Acoustic Simulation

Stephen A. Rizzi, Cochair
NASA Langley Research Center, 2 N Dryden St, MS 463, Hampton, VA 23681

Patricia Davies, Cochair
Ray W. Herrick Labs., School of Mechanical Engineering, Purdue University, 177 South Russell Street, West Lafayette, IN 47907-2099

Chair's Introduction—1:15

Invited Papers

1:20

4pNS1. Recent advances in aircraft source noise synthesis. Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., 2 N Dryden St., MS 463, Hampton, VA 23681, stephen.a.rizzi@nasa.gov), Daniel L. Palumbo (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), Jonathan R. Hardwick (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA), and Andrew Christian (National Inst. of Aerosp., Hampton, VA)

For several decades, research and development has been conducted at the NASA Langley Research Center directed at understanding human response to aircraft flyover noise. More recently, a technology development effort has focused on the simulation of aircraft flyover noise associated with future, large commercial transports. Because recordings of future aircraft are not available, the approach taken utilizes source noise predictions of engine and airframe components which serve as a basis for source noise syntheses. Human subject response studies have been conducted aimed at determining the fidelity of synthesized source noise, and the annoyance and

detectability once the noise is propagated (via simulation) to the ground. Driven by various factors, human response to less common noise sources are gaining interest. Some have been around for a long time (rotorcraft), some have come and gone, and are back again (open rotors), and some are entirely new (distributed electric driven propeller systems). Each has unique challenges associated with source noise synthesis. Discussed in this work are some of those challenges including source noise characterization from wind tunnel data, flight data, or prediction; factors affecting perceptual fidelity including tonal/broadband separation, and amplitude and frequency modulation; and a potentially expansive range of operating conditions.

1:40

4pNS2. An open architecture for auralization of dynamic soundscapes. Aric R. Aumann (Analytical Services & Mater., Inc., 107 Res. Dr., Hampton, VA 23666-1340, aric.r.aumann@nasa.gov), William L. Chapin (AuSIM, Inc., Mountain View, CA), and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

An open architecture for auralization has been developed by NASA to support research aimed at understanding human response to sound within a complex and dynamic soundscape. The NASA Auralization Framework (NAF) supersedes an earlier auralization tool set developed for aircraft flyover noise auralization and serves as a basis for a future auralization plug-in for the NASA Aircraft Noise Prediction Program (ANOPP2). It is structured as a set of building blocks in the form of dynamic link libraries, so that other soundscapes, e.g., those involving ground transportation, wind turbines, etc., and other use cases, e.g., inverse problems, may easily be accommodated The NAF allows users to access auralization capabilities in several ways. The NAF's built-in functionality may be exercised utilizing either basic (e.g., console executable) or advanced (e.g., MATLAB, LabView, etc.) host environments. The NAF's capabilities can also be extended by augmenting or replacing major activities through programming its open architecture. In this regard, it is envisioned that third parties will develop plug-in capabilities to augment those included in the NAF.

2:00

4pNS3. Simulated sound in advanced acoustic model videos. Kenneth Plotkin (Wyle, 200 12th St. South, Ste. 900, Arlington, VA 22202, kenneth.plotkin@wyle.com)

The Advanced Acoustic Model (AAM) and other time-step aircraft noise simulation models developed by Wyle can generate video animations of the noise environment. The animations are valuable for understanding details of noise footprints and for community outreach. Using algorithms developed by NASA, audio simulation for jet aircraft noise has recently been added to the video capability. Input data for the simulations consist of AAM's one-third octave band sound level time history output, plus flight path geometry and ground properties. Working at an audio sample rate of 44.1 kHz and a sample "hop" period of 0.0116 s, a random phase narrow band sample is shaped to match spectral amplitudes. Ground reflection and low frequency oscillation are added to the hops, which are merged into a WAV file. The WAV file is then mixed with an existing animation generated from the same AAM run. The process takes place in near-real time, based on a location that a user selects from a site map. The presentation includes demonstrations of the results for a simple level flyover and for the departure of a high performance jet aircraft from an airbase.

2:20

4pNS4. Combining local source propagation modeling results with a global acoustic ray tracer. Michael Williams, Darrel Younk, and Steve Mattson (Great Lakes Sound and Vib., 47140 N. Main St., Houghton, MI 49931, mikew@glsv.com)

A common method of sound auralization in large virtual environments is through acoustic ray tracing. The purpose of an acoustic ray tracer is to supply accurate source to listener impulse response functions for a virtual scene. Currently, sources are modeled as an omnidirectional point source in the ray tracer. This limits the fidelity of the results and is not accurate for complicated noise sources involving multiple audible parts. The proposed method is to simulate local source propagation to a sphere using various energy modeling techniques. These results may be used to increase the fidelity of a ray trace by giving directionality to the source and allowing for source audio to be mixed from recordings of components of the source. This is especially relevant when a full source has not yet been constructed. Because of this, there are many real world applications in engineering, architecture, and other fields that need high fidelity auralization of future products.

2:40

4pNS5. Modelling sound propagation in the presence of atmospheric turbulence for the auralization of aircraft noise. Frederik Rietdijk, Kurt Heutschi (Acoust. / Noise Control, Empa, Überlandstrasse 129, Dübendorf, Zurich 8600, Switzerland, frederik.rietdijk@empa.ch), and Jens Forssén (Appl. Acoust., Chalmers Univ. of Technol., Gothenburg, Sweden)

A new tool for the auralization of aircraft noise in an urban environment is in development. When listening to aircraft noise sound level fluctuations caused by atmospheric turbulence are clearly audible. Therefore, to create a realistic auralization of aircraft noise, atmospheric turbulence needs to be included. Due to spatial inhomogeneities of the wind velocity and temperature in the atmosphere acoustic scattering occurs, affecting the transfer function between source and receiver. Both these inhomogeneities and the aircraft position are time-dependent, and therefore the transfer function varies with time resulting in the audible fluctuations. Assuming a stationary (frozen) atmosphere, the movement of the aircraft alone gives rise to fluctuations. A simplified model describing the influence of turbulence on a moving elevated source is developed, which can then be used to simulate the influence of atmospheric turbulence in the auralization of aircraft noise.

3:00–3:20 Break

3:20

4pNS6. Simulation of excess ground attenuation for aircraft flyover noise synthesis. Brian C. Tuttle (Analytical Mech. Assoc., Inc., 1318 Wyndham Dr., Hampton, VA 23666, btuttle1@gmail.com) and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

Subjective evaluations of noise from proposed aircraft and flight operations can be performed using simulated flyover noise. Such simulations typically involve three components: generation of source noise, propagation of that noise to a receiver on or near the ground, and reproduction of that sound in a subjective test environment. Previous work by the authors focused mainly on development of high-fidelity source noise synthesis techniques and sound reproduction methods while assuming a straight-line propagation path with standard atmospheric absorption and simple (plane-wave) ground reflection models. For aircraft community noise applications, this is usually sufficient because the aircraft are nearly overhead. However, when simulating noise sources at low elevation angles, the plane-wave assumption is no longer valid and must be replaced by a model that takes into account the reflection of spherical waves from a ground surface of finite impedance. Recent additions to the NASA Community Noise Test Environment (CNoTE) software suite have improved real-time simulation capabilities of ground-plane reflections for low incidence angles. The models are presented along with the resulting frequency response of the filters representing excess ground attenuation. Discussion includes an assessment of the performance and limitations of the filters in a real-time simulation.

3:40

4pNS7. Evaluation of the perceptual fidelity of a novel rotorcraft noise synthesis technique. Jonathan R. Hardwick (Dept. of Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), Andrew Christian (National Inst. of Aerosp., 100 Exploration Way, Hampton, VA 23666, andrew.christian@nasa.gov), and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

A human subject experiment was recently conducted at the NASA Langley Research Center to evaluate the perceptual fidelity of synthesized rotorcraft source noise. The synthesis method combines the time record of a single blade passage (i.e., of a main or tail rotor) with amplitude and frequency modulations observed in recorded rotorcraft noise. Here, the single blade passage record can be determined from a time-averaged recording or from a modern aeroacoustic analysis. Since there is no predictive model available, the amplitude and frequency modulations were derived empirically from measured flyover noise. Thus, one research question was directed at determining the fidelity of four synthesis implementations (unmodulated and modulated main rotor only, and unmodulated and modulated main and tail rotor) under thickness and loading noise dominated conditions, using modulation data specific to those conditions. A second research question was aimed at understanding the sensitivity of fidelity to the choice of modulation method. In particular, can generic modulation data be used in lieu of data specific to the condition of interest, and how do modifications of generic and specific modulation data affect fidelity? The latter is of importance for applying the source noise synthesis to the simulation of complete flyover events.

4:00

4pNS8. A comparison of subjects' annoyance ratings of rotorcraft noise in two different testing environments. Andrew McMullen and Patricia Davies (Purdue Univ., 177 S Russel Dr, West Lafayette, IN 47906, almvz5@mail.missouri.edu)

Two subjective tests were conducted to investigate people's responses to rotorcraft noise. In one test subjects heard the sounds in a room designed to simulate aircraft flyovers. The frequency range of the Exterior Effects Room (EER) at NASA Langley is 17 Hz to 18,750 Hz. In the other test, subjects heard the sounds over earphones and the frequency range of the playback was 25 Hz–16 kHz. Some of the sounds in this earphone test, high-pass filtered at 25 Hz, were also played in the EER. Forty subjects participated in each of the tests. Subjects' annoyance responses in each test were highly correlated with EPNL, ASEL, and Loudness exceeded 20% of the time (correlation coefficient close to 0.9). However, at some metric values there was a large variation in response levels, which could be linked to characteristics of harmonic families present in the sound. While the results for both tests are similar, subjects in the EER generally found the sounds less annoying than the subjects who heard the sounds over earphones. Certain groups of signals were rated similarly in one test environment, but differently in the other. This may be due to playback method, subject population, or other factors.

Session 4pPA

Physical Acoustics: Topics in Physical Acoustics II

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William Slaton, Cochair
Physics & Astronomy, The University of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034

Contributed Papers

1:30

4pPA1. Aeroacoustic response of coaxial Helmholtz resonators in a low-speed wind tunnel. William Slaton (Phys. & Astronomy, The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, wvslaton@uca.edu)

The aeroacoustic response of coaxial Helmholtz resonators with different neck geometries in a low-speed wind tunnel has been investigated. Experimental test results of this system reveal strong aeroacoustic response over a Strouhal number range of 0.25–0.1 for both increasing and decreasing the flow rate in the wind tunnel. Ninety-degree bends in the resonator necks does not significantly change the aeroacoustic response of the system. Aeroacoustic response in the low-amplitude range has been successfully modeled by describing-function analysis. This analysis, coupled with a turbulent flow velocity distribution model, gives reasonable values for the location in the flow of the undulating stream velocity that drives vortex shedding at the resonator mouth. Having an estimate for the stream velocity that drives the flow-excited resonance is crucial when employing the describing-function analysis to predict aeroacoustic response of resonators.

1:45

4pPA2. Separation of acoustic waves in isentropic flow perturbations. Christian Henke (ATLAS ELEKTRONIK, Sebaldsbruecker Heerstrasse 235, Bremen 28309, Germany, christian.henke@atlas-elektronik.com)

The present contribution investigates the mechanisms of sound generation and propagation in the case of highly-unsteady flows. It is based on the linearisation of the isentropic Navier-Stokes equation around a new path-line-averaged base flow. As a consequence of this unsteady and non-radiating base flow, the perturbation equations satisfy a conservation law. It is demonstrated that this flow perturbations can be split into acoustic and vorticity modes, with the acoustic modes being independent of the vorticity modes. Moreover, we conclude that the present acoustic perturbation is propagated by the convective wave equation and fulfills Lighthill's acoustic analogy. Therefore, we can define the deviations from the convective wave equation as the "true" sound sources. In contrast to other authors, no assumptions on a slowly varying or irrotational flow are necessary.

2:00

4pPA3. The sliding mode controller on the rijke-type combustion systems with mean temperature gradients. Dan Zhao and Xinyan Li (Mech. and Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave. Singapore, Singapore 639798, Singapore, xli037@e.ntu.edu.sg)

Thermoacoustic instabilities are typically generated due to the dynamic coupling between unsteady heat release and acoustic pressure waves. To eliminate thermoacoustic instability, the coupling must be somehow interrupted. In this work, we designed and implemented a sliding mode controller to mitigate self-sustained thermoacoustic oscillations in a Rijke-type combustion system. An acoustically-compact heat source is confined and modeled by using a modified King's Law. The mean temperature gradient is considered by expanding the acoustic waves via Galerkin series. Coupling the unsteady heat release with the acoustic model enables the flow disturbances to be calculated, thus providing a platform on which to evaluate the performance of the controller. As the controller is actuated, the limit cycle oscillations are quickly dampened and the thermoacoustic system with multiple eigenmodes is stabilized. The successful demonstration indicates that the sliding mode controller can be applied to stabilize unstable thermoacoustic systems.

2:15

4pPA4. Feedback control of thermoacoustic oscillation transient growth of a premixed Laminar flame. Dan Zhao and Xy Li (Aerosp. Eng. Div., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore, XLI037@e.ntu.edu.sg)

Transient growth of combustion-excited oscillations could trigger thermoacoustic instability in a combustion system with nonorthogonal eigenmodes. In this work, feedback control of transient growth of combustion-excited oscillation in a simplified thermoacoustic system with Dirichlet boundary conditions is considered. For this a thermoacoustic model of a premixed laminar flame with an actuator is developed. It is formulated in state-space by expanding acoustic disturbances via Galerkin series and linearizing flame model and recasting it into the classical time-lag N-τ for controllers implementation. As a linear-quadratic-regulator (LQR) controller is implemented, the system becomes asymptotically stable. However, it is associated with transient growth of thermoacoustic oscillations, which may potentially trigger combustion instability. To eliminate the oscillations transient growth, a strict dissipativity controller is then implemented. Comparison is then made between the performances of these controllers. It is found that the strict dissipativity controller achieves both exponential decay of the oscillations and unity maximum transient growth.

2:30

4pPA5. Nonlinear self-sustained thermoacoustic instability in a combustor with three bifurcating branches. Dan Zhao and Shihuai Li (Aerosp. Eng. Div., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore, Singapore, LISH0025@e.ntu.edu.sg)

In this work, experimental investigations of a bifurcating thermoacoustic system are conducted first. It has a mother tube splitting into three bifurcating branches. It is surprisingly found that the flow oscillations in the bifurcating branches resulting from unsteady combustion in the bottom stem are at different temperatures. Flow visualization reveals that one branch is associated with "cold" pulsating flow, while the other two branches are "hot." Such unique flow characteristics cannot be predicted by simply assuming the bifurcating combustor consisting of three curved Rijke tube. 3D Numerical investigations are then conducted. Three parameters are identified and studied one by one: (1) the heat source location, (2) the heat flux, and (3) the flow direction in the bifurcating branches. As each of the parameters is varied, the heat-driven acoustics signature is found to change. The main nonlinearity is identified in the heat fluxes. Comparing the numerical and experimental results reveals that good agreement is obtained in terms of mode frequencies, mode shapes, sound pressure level and supercritical Hopf bifurcating behavior.

2:45

4pPA6. Application of Mach-Zehnder interferometer to measure irregular reflection of a spherically divergent N-wave from a plane surface in air. Maria M. Karzova (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Université Lyon I, Leninskie Gory 1/2, Phys. Faculty, Dept. of Acoust., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Petr V. Yuldashev (Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Sebastien Ollivier (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Université Lyon I, Lyon, France), Vera A. Khokhlova (Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Philippe Blanc-Benon (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Université Lyon I, Lyon, France)

Mach stem is a well-known structure typically observed in the process of strong (acoustical Mach numbers greater than 0.4) step-shock waves reflection from a rigid boundary. However, this phenomenon has been much less studied for weak shocks in nonlinear acoustic fields where Mach numbers are in the range from 0.001 to 0.01 and pressure waveforms have more complicated temporal structure than step shocks. In this work, the results are reported for Mach stem formation observed in the experiment of N-wave reflections from a plane surface. Spherically divergent N-waves were generated by a spark source in air and were measured using a Mach-Zehnder interferometer. Pressure waveforms were reconstructed using the inverse Abel transform applied to the phase of the interferometer measurement signal. Temporal resolution of 0.4 μs was achieved. Regular and irregular types of reflection were observed. It was shown that the length of the Mach stem increased linearly while the N-wave propagated along the surface. In addition, preliminary results of the influence of surface roughness on the Mach stem formation will be presented. [Work supported by the President of Russia MK-5895.2013.2 grant, student stipend from the French Government, and by LabEx CeLyA ANR-10-LABX-60/ANR-11-IDEX-0007.]

3:00–3:15 Break

3:15

4pPA7. Statistical inversion approach to estimating water content in an aquifer from seismic data. Timo Lähivaara (Appl. Phys., Univ. of Eastern Finland, P.O. Box 1627, Kuopio 70211, Finland, timo.lahivaara@uef.fi), Nicholas F. Dudley Ward (Otago Computational Modelling Group Ltd., Dunedin, New Zealand), Tomi Huttunen (Kuava Ltd, Kuopio, Finland), and Jari P. Kaipio (Mathematics, Univ. of Auckland, Auckland, New Zealand)

This study focuses on developing computational tools to estimate water content in an aquifer from seismic measurements. The poroelastic signature from an aquifer is simulated and methods that use this signature to estimate the water table level and aquifer thickness are investigated. In this work, the spectral-element method is used to solve the forward model that characterizes the propagation of seismic waves. The inverse problem is formulated in the Bayesian framework, so that all uncertainties are explicitly modelled as probability distributions, and the solution is given as summary statistics over the posterior distribution of parameters relative to data. For the inverse problem, we use the Bayesian approximation error method which reduces the overall computational demand. In this study, results in the two-dimensional case with simulated data are presented.

3:30

4pPA8. Surfactant-free emulsification in microfluidics using strongly oscillating bubbles. Siew-Wan Ohl, Tandiono Tandiono, Evert Klaseboer (Inst. of High Performance Computing, 1 Fusionopolis Way, #16-16 Connexis North, Singapore 138632, Singapore, ohlsw@ihpc.a-star.edu.sg), Dave Ow, Andre Choo (Bioprocessing Technol. Inst., Singapore, Singapore), Fenfang Li, and Claus-Dieter Ohl (Division of Phys. and Appl. Phys., School of Physical and Mathematical Sci., Nanyang Technol. Univ., Singapore, Singapore)

In this study, two immiscible liquids in a microfluidics channel has been successfully emulsified by acoustic cavitation bubbles. These bubbles are generated by the attached piezo transducers which are driven to oscillate at resonant frequency of the system (about 100 kHz) [1, 2]. The bubbles oscillate and induce strong mixing in the microchamber. They induce the rupture of the liquid thin layer along the bubble surface due to the high shear stress and fast liquid jetting at the interface. Also, they cause the big droplets to fragment into small droplets. Both water-in-oil and oil-in-water emulsions with viscosity ratio up to 1000 have been produced using this method without the application of surfactant. The system is highly efficient as submicron monodisperse emulsions (especially for water-in-oil emulsion) could be created within milliseconds. It is found that with a longer ultrasound exposure, the size of the droplets in the emulsions decreases, and the uniformity of the emulsion increases. Reference: [1] Tandiono, SW Ohl *et al.*, "Creation of cavitation activity in a microfluidics device through acoustically driven capillary waves," Lab Chip 10, 1848–1855 (2010). [2] Tandiono, SW Ohl *et al.*, "Sonochemistry and sonoluminescence in microfluidics," Proc. Natl. Acad. Sci. U.S.A. 108(15), 5996–5998 (2011).

3:45

4pPA9. Ultrasonic scattering from poroelastic materials using a mixed displacement-pressure formulation. Max denis (Mayo Clinic, 200 First St. SW, Rochester, MN 55905, denis.max@mayo.edu), Chrisna Nguon, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, Lowell, MD)

In this work, a numerical technique suitable for evaluating the ultrasonic scattering from a three-dimensional poroelastic material is presented. Following Biot's derivation of the macroscopic governing equations for a fluid saturated poroelastic material, the predicted two propagating wave equations are formulated in terms of displacement and pressure. Assuming that porosity variations on a microscopic scale have a cumulative effect in generating a scattered field, the scattering attenuation coefficient of a Biot medium can be determined. The scattered fields of the wave equations are numerically evaluated as Neumann series solutions of the Kirchhoff-Helmholtz integral equation. A Padé approximant technique is employed to extrapolate beyond the Neumann series' radius of convergence (weak scattering regime). In the case of bovine trabecular bone, the relationship between the scattering attenuation coefficient and the structural and mechanical properties of the trabecular bone is of particular interest. The results demonstrate the validity of the linear frequency-dependent assumption of attenuation coefficient n the low frequency range. Further comparisons, between measured observations and the numerical results will be discussed.

4:00

4pPA10. High temperature resonant ultrasound spectroscopy study on Lead Magnesium Niobate—Lead Titanate (PMN-PT) relaxor ferroelectric material. Sumudu P. Tennakoon and Joseph R. Gladden (Phys. and Astronomy, Univ. of MS, 1 Coliseum Dr., Phys.& NCPA, Univ. of MS, University, MS 38677, sptennak@go.olemiss.edu)

Lead magnesium niobate-lead titanate [(1-x)$PbMg_{1/3}Nb_{2/3}O_3$-x$PbTiO_3$] is a perovskite relaxor ferroelectric material exhibiting superior electromechanical coupling compared to the conventional piezoelectric materials. In this work, non-poled single crystal PMN-PT material with the composition near morphotropic phase boundary (MPB) was investigated in the temperature range of 400 K—800 K where the material is reported to be in the cubic phase. High temperature resonant ultrasound spectroscopy (HT-RUS) technique was used to probe temperature dependency of elastic constants derived from the measured resonant modes. Non-monotonic resonant frequency trends in the temperature regime indicate stiffening of the material, followed by gradual softening typically observed in heated materials. Elastic constants confirmed this stiffening in the temperature range of 400 K—673 K, where the stiffness constants C_{11} and C_{44} increased approximately by 40% and 33% respectively. Acoustic attenuation, derived from the quality factor (Q), exhibits a minimum around the temperature where the stiffness is maximum and, significantly higher attenuation observed at temperatures below 400 K. The temperature range 395 K—405 K was identified as a transition temperature range, where the material showed an abrupt change in the resonant spectrum and, the material emerges from the MPB characterized by this very high acoustic attenuation. This transition temperature compares favorably with dielectric constant measurements reported in the literature.

4:15

4pPA11. Structure of cavitation zones in a heavy magma under explosive character of its decompression. Valeriy Kedrinskiy (Physical HydroDynam.., Lavrentyev Inst. of HydroDynam.., Russian Acad. of Sci., Lavrentyev prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru)

The paper is devoted to the investigation of a dynamics of state and structure of compressed magma flow saturated by gas and microcrystallites which is characterized by phase transitions, diffusive processes, by increase of a magma viscosity magnitude by the orders and bubbly cavitation development behind the decompression wave front formed in the result of volcanic channel depressurization. The multi-phase mathematical model, which includes well-known conservation laws for mean pressure, mass velocity, and density as well as the system of the kinetic equations describing the physical processes that occur in a compressed magma during its explosive decompression, is considered. The results of numerical analysis show that the processes of a magma saturation by cavitation nuclei as their density magnitude increases by a few orders lead to the formation of separate zone with anomalously high values of the flow parameters. As it has turned out the abnormal zone is located in the vicinity of a free surface of a cavitating magma column. The mechanism of its formation is determined by diffusion flows redistribution as the nuclei density increases as well as by the change of the distribution character of main flow parameters in the abnormal zone from a gradual to an abrupt increase of their values on the lower zone bound. Note, the mass velocity jump by the order magnitude relatively main flow allows to conclude that the flow disintegration on the lower bound of the zone is quite probable. [Supp. RAS Presidium Program, Project 2.6].

4:30

4pPA12. Cavity collapse in a bubbly liquid. Ekaterina S. Bolshakova (Phys., Novosibirsk State Univ., Novosibirsk, Russian Federation) and Valeriy Kedrinskiy (Physical HydroDynam., Lavrentyev Inst. of HydroDynam., Russian Acad. of Sci., Lavrentyev prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru)

The effect of an ambient liquid state on a spherical cavity dynamics under atmospheric hydrostatic pressure p and extremely low initial gas pressure p(0) inside was investigated. The equilibrium bubbly (cavitating) medium with sound velocity C as the function of gas phase concentration k was considered as the ambient liquid model. The cavity dynamics is analyzed within the framework of Herring-equation for the following diapasons of main parameters : k = 0—5%, p(0) = 0.02—10(−6) atm. Numerical analysis has shown that the deceleration C by two order does not have an influence neither on an asymptotic value of collapsed cavity radius nor on the acoustical losses under its collapse. It means than in the whole the integral acoustical losses remain invariable. However the collapse cavity dynamics and the radiation structure are essentially changed: from numerous pulsations with a decreasing amplitudes up to a single collapse and from a wave packet up to a single wave, correspondingly. It has turned out that the acoustic corrections in the Herring-equation don't influence practically on the cavity dynamics if the term of equation with dH/dt is absent. Naturally, the deceleration C exerts essential influence on an empty cavity dynamics. The graphs of dR/Cdt values as a function of R/Ro for different C values are located higher the data of classical models of Herring, Gilmore and Hunter. So the value M = 1 is reached at R/Ro = 0.023 for k = 0, and at the value 0.23 when k = 5 %. [Support RFBR, grant 12-01-00314.]

Session 4pPP

Psychological and Physiological Acoustics: Physiological and Psychological Aspects of Central Auditory Processing Dysfunction II

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Adrian KC Lee, Cochair
University of Washington, Box 357988, University of Washington, Seattle, WA 98195

Invited Papers

1:30

4pPP1. Aging as a window into central auditory dysfunction: Combining behavioral and electrophysiological approaches. David A. Eddins, Erol J. Ozmeral, and Ann C. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu)

Central auditory processing involves a complex set of feed-forward and feedback processes governed by a cascade of dynamic neuro-chemical mechanisms. Central auditory dysfunction can arise from disruption of one or more of these processes. With hearing loss, dysfunction may begin with reduced and/or altered input to the central auditory system followed by peripherally induced central plasticity. Similar central changes may occur with advancing age and neurological disorders even in the absence of hearing loss. Understanding the behavioral and physiological consequences of this plasticity on the processing of basic acoustic features is critical for effective clinical management. Major central auditory processing deficits include reduced temporal processing, impaired binaural hearing, and altered coding of spectro-temporal features. These basic deficits are thought to be primary contributing factors to the common complaint of difficulty understanding speech in noisy environments in persons with hearing loss, brain injury, and advanced age. The results of investigations of temporal, spectral, and spectro-temporal processing, binaural hearing, and loudness perception will be presented with a focus on central auditory deficits that occur with advancing age and hearing loss. Such deficits can be tied to reduced peripheral input, altered central coding, and complex changes in cortical representations.

2:00

4pPP2. Age-related declines in hemispheric asymmetry as revealed in the binaural interaction component. Ann C. Eddins, Erol J. Ozmeral, and David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, aeddins@usf.edu)

The binaural interaction component (BIC) is a physiological index of binaural processing. The BIC is defined as the brain activity resulting from binaural (diotic or dichotic) stimulus presentation minus the brain activity summed across successive monaural stimulus presentations. Smaller binaural-induced activity relative to summed monaural activity is thought to reflect neural inhibition in the central auditory pathway. Since aging is commonly associated with reduced inhibitory processes, we evaluate the hypothesis that the BIC is reduced with increasing age. Furthermore, older listeners typically have reduced hemispheric asymmetry relative to younger listeners, interpreted in terms of compensation or recruitment of neural resources and considered an indication of age-related neural plasticity. Binaural stimuli designed to elicit a lateralized percept generate maximum neural activity in the hemisphere opposite the lateralized position. In this investigation, we evaluated the hypothesis that the BIC resulting from stimuli lateralized to one side (due to interaural time differences) results in less hemispheric asymmetry in older than younger listeners with normal hearing. Behavioral data were obtained to assess the acuity of binaural processing. Data support the interpretation that aging is marked by reduced central auditory inhibition, reduced temporal processing, and broader distribution of activity across hemispheres compared to young adults.

2:30

4pPP3. Effects of blast exposure on central auditory processing. Melissa Papesh, Frederick Gallun, Robert Folmer, Michele Hutter, M. Samantha Lewis, Heather Belding (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 6303 SW 60th Ave., Portland, OR 97221, Melissa.Papesh@va.gov), and Marjorie Leek (Res., Loma Linda VA Medical Ctr., Loma Linda, CA)

Exposure to high-intensity blasts is the most common cause of injury in recent U.S. military conflicts. Prior work indicates that blast-exposed Veterans report significantly more hearing handicap than non-blast-exposed Veterans, often in spite of clinically normal hearing thresholds. Our research explores the auditory effects of blast exposure using a combination of self-report, behavioral, and electrophysiological measures of auditory processing. Results of these studies clearly indicate that blast-exposed individuals are significantly more likely to perform poorly on tests requiring the use of binaural information and tests of pitch sequencing and temporal acuity

compared to non-blast-exposed control subjects. Behavioral measures are corroborated by numerous objective electrophysiological measures, and are not necessarily attributable to peripheral hearing loss or impaired cognition. Thus, evidence indicates that blast exposure can lead to acquired deficits in central auditory processing (CAP) which may persist for at least 10 years following blast exposure. Future studies of these deficits in this and other adult populations are needed to address important issues such as individual susceptibility, anatomical, and physiological changes in auditory pathways which contribute to symptoms of these types of deficits, and development of effective evidence-based methods of rehabilitation in adult patients. [Work supported by the VA Rehabilitation Research & Development Service and the VA Office of Academic Affiliations.]

3:00–3:30 Break

3:30

4pPP4. Auditory processing demands and working memory span. Margaret K. Pichora-Fuller (Dept. of Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, k.pichora.fuller@utoronto.ca) and Sherri L. Smith (Audiologic Rehabilitation Lab., Veterans Affairs, Mountain Home, TN)

The (in)dependence of auditory and cognitive processing abilities is a controversial topic for hearing researchers and clinicians. Some advocate for the need to isolate auditory and cognitive factors. In contrast, we argue for the need to understand how they interact. Working memory span (WMS) is a cognitive measure that has been related to language comprehension in general and also to speech understanding in noise. In healthy adults with normal hearing, there is typically a strong correlation between reading and listening measures of WMS. Some investigators have opted to use visually presented stimuli when testing people who do not have normal hearing in order to avoid the influence of modality-specific auditory processing deficits on WMS. However, tests conducted using auditory stimuli are necessary to evaluate how cognitive processing is affected by the auditory processing demands experienced by different individuals over a range of conditions in which the tasks to be performed, the availability of supportive context, and the acoustical and linguistic characteristics of targets and maskers are varied. Attempts to measure auditory processing independent of cognitive processing will fall short in assessing listening function in realistic conditions.

4:00

4pPP5. Auditory perceptual learning as a gateway to rehabilitation. Beverly A. Wright (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, b-wright@northwestern.edu)

A crucial aspect of the central nervous system is that it can be modified through experience. Such changes are thought to occur in two learning phases: acquisition—the actual period of training—and consolidation—a post-training period during which the acquired information is transferred to long-term memory. My coworkers and I have been addressing these principles in auditory perceptual learning by characterizing the factors that induce and those that prevent learning during the acquisition and consolidation phases. We also have been examining how these factors change during development and aging and are affected by hearing loss and other conditions that alter auditory perception. Application of these principles could improve clinical training strategies. Further, though learning is the foundation for behavioral rehabilitation, the capacity to learn can itself be impaired. Therefore, an individual's response to perceptual training could be used as an objective, clinical measure to guide diagnosis and treatment of a cognitive disorder. [Work supported by NIH.]

4:30–5:00 Panel Discussion

Session 4pSC

Speech Communication: Voice (Poster Session)

Richard J. Morris, Chair
Communication Science and Disorders, Florida State University, 201 West Bloxham Road, 612 Warren Building, Tallahassee, FL 32306-1200

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors an opportunity to see other posters, the contributors of odd-numbered papers will be at their posters from 1:00 p.m. and 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

4pSC1. Acoustical bases for the perception of simulated laryngeal vocal tremor. Rosemary A. Lester, Brad H. Story, and Andrew J. Lotto (Speech, Lang., and Hearing Sci., Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721, rosemary.lester@gmail.com)

Vocal tremor involves atypical modulation of the fundamental frequency (F0) and intensity of the voice. Previous research on vocal tremor has focused on measuring the modulation rate and extent of the F0 and intensity without characterizing other modulations present in the acoustic signal (i.e., modulation of the harmonics). Characteristics of the voice source and vocal tract filter are known to affect the amplitude of the harmonics and could potentially be manipulated to reduce the perception of vocal tremor. The purpose of this study was to determine the adjustments that could be made to the voice source or vocal tract filter to alter the acoustic output and reduce the perception of modulation. This research was carried out using a computational model of speech production that allows for precise control and modulation of the glottal and vocal tract configurations. Results revealed that listeners perceived a higher magnitude of voice modulation when simulated samples had a higher mean F0, greater degree of vocal fold adduction, and vocal tract shape for /i/ vs. /ɑ/. Based on regression analyses, listeners' judgments were predicted by modulation information present in both low and high frequency bands. [Work supported by NIH F31-DC012697.]

4pSC2. Perception of breathiness in pediatric speakers. Lisa M. Kopf, Rahul Shrivastav (Communicative Sci. and Disord., Michigan State Univ., Rm. 109, Oyer Speech and Hearing Bldg., 1026 Red Cedar Rd., East Lansing, MI 48824, kopflisa@msu.edu), David A. Eddins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), and Mark D. Skowronski (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Extensive research has been done to determine acoustic metrics for voice quality. However, few studies have focused on voice quality in the pediatric population. Instead, metrics evaluated on adults have directly been applied to children's voices. Some variables, such as pitch, that differ between adult and pediatric voices, have been shown to be critical in the perception of breathiness. Furthermore, it is not known whether adults perceive voice quality similarly for pediatric and adult speakers. In this experiment, 10 listeners judged breathiness for 28 stimuli using a single-variable matching task. The stimuli were modeled after four pediatric speakers and synthesized using a Klatt-synthesizer to have a wide range of aspiration noise and open quotient. Both of these variables have been shown to influence the perception of breathiness. The resulting data were compared to that previously obtained for adult speakers using the same matching task. Comparison of adult and pediatric voices will help identify differences in the perception of breathiness for these groups of speakers and to develop more accurate metrics for voice quality in children. [Research supported by NIH (R01 DC009029).]

1:00

4pSC3. Combining differentiated electroglottograph and differentiated audio signals to reliably measure vocal fold closed quotient. Richard J. Morris (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu), Shonda Bernadin (Elec. and Comput. Eng., Florida A & M Univ., Tallahassee, FL), David Okerlund (College of Music, Florida State Univ., Tallahassee, FL), and Lindsay B. Wright (Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL)

Over the past few decades researchers have explored the use of the electroglottograph (EGG) as a non-invasive method for representing vocal fold contact during vowel production and to measure the closed quotient (CQ) and open quotient (OQ) of the glottal cycle. The first derivative of the EGG signal (dEGG) can be used to indicate these moments (Childers & Krishnamurthy, 1985). However, there can be double positive peaks in the dEGG as well as a variety of negative peak patterns (Herbst *et al*., 2010). Obviously these variations will alter any measurements made from the signal. Recently, the use of the dEGG with dAudio signal was reported as a means for more reliable measurement of the CQ from the EGG signal in combination with a time synchronized audio signal. The purpose of this study is to demonstrate the reliability of the dEGG and dAudio for determining CQ across a variety of vocal conditions. Files recorded from group of 15 trained females singing an octave that included their primo passaggio provided the data. Preliminary results indicate high reliability of the CQ measurements in both the chest and middle registers of all of the singers.

4pSC4. A reduced-order three-dimensional continuum model of voice production. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Although vocal fold vibration largely occurs in the transverse plane, control of voice is mainly achieved by adjusting vocal fold stiffness along the anterior–posterior direction through muscle activation. Thus, models of voice control need to be at least three-dimensional on the structural side. Modeling the detailed three-dimensional interaction between vocal fold dynamics, glottal aerodynamics, and the sub- and supra-glottal acoustics is computationally expensive, which prevents parametric studies of voice production using three-dimensional models. In this study, a Galerkin-based reduced-order three-dimensional continuum model of phonation was presented. Preliminary results showed that this model was able to qualitatively reproduce previous experimental observations. This model is computationally efficient and thus ideal for parametric studies in phonation research as well as practical applications such as speech synthesis. [Work supported by NIH.]

4pSC5. The influence of attentional focus on voice control. Eesha A. Zaher and Charles R. Larson (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, EeshaZaheer2014@u.northwestern.edu)

The present study tested the role of attentional focus on control of voice fundamental frequency (F0). Subjects vocalized an "ah" sound while hearing their voice auditory feedback randomly shifted upwards or downwards in pitch. In the "UP" condition, subjects vocalized, listened for and pressed a button for each upward pitch shift stimulus. In the "DOWN" condition, subjects listened for and pressed a button for each downward shift. In the CONTROL condition, subjects vocalized without paying attention to the stimulus direction or pressing a button. Data were analyzed by averaging voice F0 contours across several trials for each pitch shift stimulus in all conditions. Response magnitudes were larger for the CONTROL than for the UP or DOWN conditions. Responses for the UP and DOWN conditions did not differ. Results suggest that when subjects focus their attention to identify specific stimuli and produce a non-vocal motor response conditional upon the identification, the neural mechanisms involved in voice control are reduced, possibly because of a reduction in the error signal resulting from the comparison of the efference copy of voice output with auditory feedback. Thus, focusing attention away from vocal control reduces neural resources involved in control of voice F0.

4pSC6. Attention-related modulation of involuntary audio-vocal response to pitch feedback errors. Hanjun Liu, Huijing Hu, and Ying Liu (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

It has been demonstrated that unexpected alterations in auditory feedback elicit fast compensatory adjustments in vocal production. Although generally thought to be involuntary in nature, whether these adjustments can be influenced by cognitive function such as attention remains unknown. The present event-related potential (ERP) study investigated whether neurobehavioral processing of auditory-vocal integration can be affected by attention. While sustaining a vowel phonation and hearing pitch-shifted feedback, participants were required to either ignore the auditory feedback perturbation, or attend to it with two levels of attention load. The results revealed enhancement of P2 response to the attended auditory perturbation with the low load level as compared to the unattended auditory perturbation. Moreover, increased auditory attention load led to a significant decrease of P2 response. By contrast, there was no attention-related change of vocal response. These findings provide the first neurophysiological evidence that involuntary auditory-vocal integration can be modulated as a function of auditory attention. Furthermore, it is suggested that auditory attention load can result in a decrease of the cortical processing of auditory-vocal integration in pitch regulation.

4pSC7. A study on the effect of intraglottal vortical structures on vocal fold vibration. Mehrdad H Farahani and Zhaoyan Zhang (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., UCLA School of Medicine, 1000 Veteran Ave., Los Angeles, CA 90095, mh.farahani@gmail.com)

Recent investigations suggested possible formation of the vortical structures in the intraglottal region during the closing phase of the phonation cycle. Vortical regions in the flow field are locations of negative pressure, and it has been hypothesized that this negative pressure might facilitate the glottal closure and thus affects the vibration pattern and voice production for high subglottal pressures. However, it is unclear whether the vortex-induced negative pressure is large enough, compared with vocal fold inertia and elastic recoil, to have a noticeable effect on glottal closure. In addition, the intraglottal vortical structures generally exist only for a small fraction of the closing phase when the glottis becomes divergent enough to induce flow separation. In the current work, oscillation of the vocal folds and the flow field are modeled using a non-linear finite element solver and a reduced order flow solver, respectively. The effect of vortical structures is modeled as a sinusoidal negative pressure wave applied to vocal fold surface between the flow separation point and the superior edge of the vocal folds. The effects of this vortex-induced negative pressure are quantified at different conditions of vocal fold stiffness and subglottal pressures. [Work supported by NIH.]

4pSC8. Effects of thyroarytenoid muscle activation on phonation in an *in vivo* canine larynx model. Georg Luegmair, Dinesh Chhetri, and Zhaoyan Zhang (Dept. of Head and Neck Surgery, Univ. of California Los Angeles, 1000 Veteran Ave., Rehab 31-24, Los Angeles, CA 90095, gluegmair@ucla.edu)

Previous studies have shown that the thyroarytenoid (TA) muscle plays an important role in the control of vocal fold adduction and stiffness. The effects of TA stimulation on vocal fold vibration, however, are still unclear. In this study, the effects of TA muscle activation on phonation were investigated in an *in vivo* canine larynx model. Laryngeal muscle activation was achieved through parametric stimulation of the thyroarytenoid, the lateral cricoarytenoid (LCA), and the cricothyroid (CT) muscles. For each stimulation level, the subglottal pressure was gradually increased to produce phonation. The subglottal pressure, the volume flow, and the outside acoustic pressure were measured together with high-speed recording of vocal fold vibration from a superior view. The results show that, without TA activation, phonation was limited to conditions of medium to high levels of LCA and CT activations. TA activation allowed phonation to occur at a much lower activation level of the LCA and CT muscles. Compared to conditions of no TA activation, TA activation led to decreased open quotient. Increasing TA activation also allow phonation to occur at a much larger range of the subglottal pressure while still maintaining certain degree of glottal closure during vibration. [Work supported by NIH.]

4pSC9. Voice accumulation and voice disorders in primary school teachers. Pasquale Bottalico (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 10125, pasqualebottalico@yahoo.it), Lorenzo Pavese, Arianna Astolfi (Dipartimento di Energia, Politecnico di Torino, Torino, Italy), and Eric J. Hunter (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Statistics on professional voice users with vocal health issues demonstrate the significance of the problem. However, such disorders are not currently recognized as an occupational disease in Italy. Conducting studies examining the vocal health of occupational voice users is an important step in identifying this as an important public health issue. The current study was conducted in six primary schools in Italy with 25 teachers, one of the most affected occupational categories. A clinical examination was conducted (consisting of hearing and voice screening, a VHI, etc.). On this basis, teachers were divided into three groups: healthy subjects, subject with logopaedic disorders, and subjects with objectively measured pathological symptoms. The distributions of voicing and silence periods for the teachers at work were collected using the Ambulatory Phonation Monitor (APM3200), a device for long-term monitoring of vocal parameters. The APM senses the vocal fold vibrations at the base of the neck by means of a small accelerometer. Correlations were calculated between the voice accumulation slope (obtained by multiplying the number of occurrences for each period by the corresponding duration) and the clinical status of the teachers. The differences in voice accumulation distributions among the three groups were analyzed.

4pSC10. Room acoustics and vocal comfort in untrained vocalists. Eric J. Hunter, Pasquale Bottalico, Simone Graetzer, and Russell Banks (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ejhunter@msu.edu)

Talkers have long been studied in their speech accommodation strategies in noise. Vocal effort and comfort within noisy situations have also been studied. In this study, untrained vocalists were exposed to a range of room acoustic conditions. In each environment, the subject performed a vocal task, with a goal of being "heard" by a listener 5 m away. After each task, the subject completed a series of questions addressing vocal effort and comfort. Additionally, using a head and torso simulator (HATS), the environment was assessed using a sine sweep presented at the HATS mouth and recorded at the ears. It was found that vocal clarity (C50) and the initial reflection related to vocal comfort. The results are not only relevant to room design but also to understanding talkers' acuity to acoustic conditions and their adjustments to them.

4pSC11. Flow vibrato in singers. Srihimaja Nandamudi and Ronald C. Scherer (Commun. Sci. and Disord., Bowling Green State Univ., 200 Health and Human Services Bldg., Bowling Green, OH 43403, nandas@bgsu.edu)

Frequency (F0) vibrato is commonly known, but not so for flow vibrato, the mean flow variation that accompanies frequency vibrato. Two classically trained singers, each with over 20 years professional experience, a soprano and a tenor, recorded /pa:pa:pa:/ sequences on three pitches (C4, A4, and G5 for the soprano, D3, D4, and G4 for the tenor) and three loudness levels (p, mf, and f) at each pitch. Each vowel had 3–6 frequency vibrato cycles. For both singers, flow vibrato (obtained using the Glottal Enterprises aerodynamic system) was present, and the lowest pitch had the most variability; otherwise, flow vibrato was fairly sinusoidal in shape. For the soprano, flow vibrato cycle extents were: 21–88 cc/s, lowest pitch; 60–147 cc/s, middle pitch; 115–214 cc/s, highest pitch, across loudness levels. For the soprano, the phase difference for the flow was 120–180 degrees ahead of the F0 vibrato. For the tenor, the flow vibrato cycle extents were: 32–85 cc/s, lowest pitch; 98–113 cc/s, middle pitch; 76–240 cc/s, highest pitch, across loudness levels. Flow vibrato for the tenor led the F0 vibrato typically by 40–120 degrees. For both subjects, some flow vibrato cycles had double peaks. Flow vibrato needs further study to determine its origin, shapes, and magnitudes.

4pSC12. Impact of vocal tract resonance on the perception of voice quality changes caused by vocal fold stiffness. Rosario Signorello, Zhaoyan Zhang, Bruce Gerratt, and Jody Kreiman (Head and Neck Surgery, Univ. of California Los Angeles David Geffen School of Medicine, 31-24 Rehab Ctr., UCLA School of Medicine, 1000 Veteran Ave., Los Angeles, CA 90095, rsignorello@ucla.edu)

Experiments using animal and human larynx models are often conducted without a vocal tract. While it is reasonable to assume the absence of a vocal tract has only small effects on vocal fold vibration, it is unclear how sound production and its perception will be affected. In this study, the validity of using data obtained in the absence of a vocal tract for voice perception studies was investigated. Using a two-layer self-oscillating physical model, three series of voice stimuli were created: one produced with conditions of left-right symmetric vocal fold stiffness, and two with left-right asymmetries in vocal fold body stiffness. Each series included a set of stimuli created with a physical vocal tract, and a second set created without a physical vocal tract. Stimuli were re-synthesized to equalize the mean F0 for each series and normalized for amplitude. Listeners were asked to evaluate the three series in a sort-and-rate task. Multidimensional scaling analysis will be applied to examine the perceptual interaction between the voice source and the vocal tract resonances. [Work supported by NIH.]

4pSC13. Perceptual differences among models of the voice source: Further evidence. Marc Garellek (Linguist, UCSD, La Jolla, CA), Gang Chen (Elec. Eng., UCLA, Los Angeles, CA), Bruce R. Gerratt (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403), Abeer Alwan (Elec. Eng., UCLA, Los Angeles, CA), and Jody Kreiman (Head and Neck Surgery, UCLA, Los Angeles, CA, jkreiman@ucla.edu)

Models of the voice source differ in how they fit natural voices, but it is still unclear which differences in fit are perceptually salient. This study describes ongoing analyses of differences in the fit of six voice source models to 40 natural voices, and how these differences relate to perceptual similarities among stimuli. Listeners completed a visual sort-and-rate task to compare versions of each voice created with the different source models, and the results were analyzed using multi-dimensional scaling (MDS). Perceptual spaces were interpreted in terms of variations in model fit in both the time and spectral domains. The discussion will focus on the perceptual importance of matches to both time-domain and spectral features of the voice. [Work supported by NIH/NIDCD grant DC01797 and NSF grant IIS-1018863.]

4pSC14. The biological function of fundamental frequency in leaders' charismatic voices. Rosario Signorello (Head and Neck Surgery, Univ. of California Los Angeles David Geffen School of Medicine, 31-24 Rehab Ctr., UCLA School of Medicine, 1000 Veteran Ave., Los Angeles, CA 90095, rsignorello@ucla.edu)

Charismatic leaders use voice based on two functions: a primary biological function and a secondary language and culture-based function (Signorello, 2014). In order to study the primary function in more depth, we conducted acoustic and perceptual studies on the use of F0 by French, Italian and Brazilian charismatic political leaders. Results show that leaders manipulate F0 in significantly different manners relative to: (1) the context of communication (persuasive goal, the place where communication occurs and the type of audience) in order to be recognized as the leader of the group; and (2) the elapse of time (from the beginning to the end of the speech) in order to create a climax with the audience. Results of a perceptual test show that the leader's use of low F0 voice results in the perception of the leader as a dominant or threatening leader and the use of higher F0 conveys sincere, calm, and reassuring leadership. These results show cross-language and cross-cultural similarities in leaders' vocal behavior and listeners' perception, and robustly demonstrate the two different functions of leaders' voices.

4pSC15. Voice quality variation and gender. Kara Becker, Sameer ud Dowla Khan, and Lal Zimman (Linguist, Reed College, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, kbecker@reed.edu)

Recent work on American English has established that speakers increasingly use creaky phonation to convey pragmatic information, with young urban women assumed to be the most active users of this phonetic feature. However, no large-scale acoustic or articulatory study has established the actual range and diversity of voice quality variation along gender identities, encompassing different sexual orientations, regional backgrounds, and socioeconomic statuses. The current study does exactly that, through four methods: (1) subjects identifying with a range of gender and other demographic identities were audio recorded while reading wordlists as well as a scripted narrative assuming characters' voices designed to elicit variation in vowel quality. Simultaneously, (2) electroglottographic readings were taken and analyzed to determine the glottal characteristics of this voice quality variation. (3) Subjects were then asked to rate recordings of other people's voices to identify the personal characteristics associated with the acoustic reflexes of phonation; in the final task, (4) subjects were explicitly asked about their language ideologies as they relate to gender. Thus, the current study explores the relation between gender identity and phonetic features, measured acoustically, articulatorily, and perceptually. This work is currently underway and preliminary results are being compiled at this time.

4pSC16. Towards standard scales for dysphonic voice quality: Magnitude estimation of reference stimuli. David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu) and Rahul Shrivastav (Communicative Sci. & Disord., Michigan State Univ., East Lansing, MI)

This work represents a critical step in developing standard measurement scales for the dysphonic voice qualities of breathiness and roughness. Methods such as Likert ratings, visual analog scales and magnitude estimation result in arbitrary units, limiting their clinical usefulness. A single-variable matching task can quantifying voice quality in terms of physical units but is too time consuming for clinical use. None of these methods result in information that has a direct or intuitive relationship with the underlying percept. A proven approach for the perception of loudness is the Sone scale which ties physical units to the perceptual estimates of loudness magnitude. As a first step in developing such a scale for breathiness and roughness, here we establish the relationship between the change in perceived VQ magnitude and the change in physical units along the continuum of each VQ dimension. A group of 25 listeners engaged in a magnitude estimation task to determine perceived magnitude associated with the comparison stimuli used in our single-variable matching tasks. This relationship is analogous to mapping intensity in dB to perceived loudness in Phons and is a critical step in developing a Sone-like scale for breathiness and roughness.

4pSC17. Divergent or convergent glottal angles: Which give greater flow? Ronald Scherer (Commun. Sci. and Disord., Bowling Green State Univ., 200 Health Ctr., Bowling Green, OH 43403, ronalds@bgsu.edu)

During phonation, the glottis alters between convergent and divergent angles. For the same angle value, diameter, and transglottal pressure, which angle, divergent or convergent, results in greater flow? The symmetric glottal angles of the physical static model M5 were used. Characteristics (life-size) of the model were: axial glottal length 0.30 cm; angles of 5, 10, 20, and 40 degrees; diameters of 0.005, 0.01, 0.02, 0.04, 0.08, 0.16, and 0.32 cm; transglottal pressures from 1 to 25 cm H2O; resulting in flows from 2.7 to 1536 cc/s and Reynolds number from 29.4 to 13,058. Results: (1) For diameters of 0.04, 0.08 and 0.16 cm, the divergent angle always gave more flow than the convergent angle (about 5–25%); (2) for the smallest (0.005 cm) and largest diameter (0.32 cm), the divergent angles always gave less flow (10–30%); (3) for diameters of 0.01 and 0.02 cm, flow was greater for divergent 5 and 10 degrees, and less for divergent 20 and 40 degrees. These results suggest that the divergent glottal angle will increase the glottal flow for midrange glottal diameters (skewing the glottal flow further "to the right"?), and create less flow at very small diameters (increasing energy in the higher harmonics?).

4pSC18. Methodological issues when estimating subglottal pressure from oral pressure. Brittany Frazer (Commun. Sci. and Disord., Bowling Green State Univ., 200 Marie Pl., Perrysburg, OH 43551, bfrazer@bgsu.edu) and Ronald C. Scherer (Commun. Sci. and Disord., Bowling Green State Univ., Bowling Green, OH)

A noninvasive method to estimate subglottal pressure for vowel productions is to smoothly utter a CVCV string such as /p:i:p:i:p:i:…/ using a short tube in the mouth with the tube attached to a pressure transducer. The pressure during the lip occlusion estimates the subglottal pressure during the adjacent vowel. What should the oral pressure look like for it to provide accurate estimates? The study compared results using various conditions against a standard condition that required participants to produce /p:i:p:i:../ syllables smoothly and evenly at approximately 1.5 syllables per second. The non-standard tasks were: performing the task without training, increasing syllable rate, using a voiced /b/ instead of a voiceless /p/ initial syllable, adding a lip or velar leak, or using a two syllable production ("peeper") instead of a single syllable production. Lip leak, velar leak, and lack of time to equilibrate air pressure throughout the airway caused estimates of subglottal pressure to be inaccurate. Accuracy was better when estimates of subglottal pressure were obtained using the voiced initial consonant and the two-syllable word. Training improved the consistency of the oral pressure profiles and thus the assurance in estimating the subglottal pressure. Oral pressures with flat plateaus appear most accurate.

THURSDAY AFTERNOON, 30 OCTOBER 2014 — INDIANA F, 1:00 P.M. TO 4:45 P.M.

Session 4pUW

Underwater Acoustics: Shallow Water Reverberation III

Kevin L. Williams, Chair
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Contributed Paper

1:00

4pUW1. Seafloor sound-speed profile and interface dip angle measurement by the image source method: Application to real data. Samuel Pinson (Laboratório de Vibrações e Acústica, Universidade Federal de Santa Catarina, LVA Dept de Engenharia Mecânica, UFSC, Bairro Trindade, Florianópolis, SC 88040-900, Brazil, samuelpinson@yahoo.fr) and Charles W. Holland (Penn State Univ., State College, PA)

The image source method characterizes the sediment sound-speed profile from seafloor reflection data with a low computational cost compared with inversion techniques. Recently, the method has been extended to treat non-parallel sediment layering. The method is applied to data from an autonomous underwater vehicle (AUV) towing a source (1600–3500 Hz) and an horizontal array of hydrophones. AUV reflection measurements were acquired every 3 m along 10 criss-cross lines over a 1km<+>2<+> area with evidently dipping layers. Mapping the along track sound-speed profiles in geographical coordinates results in a pseudo-3D (Nx2D) sediment structure characterization of the area down to several tens of meters in the sub-bottom. The sound speed profile agreement at crossing points is quite good.

Invited Papers

1:15

4pUW2. Requirements, technology, and science drivers of applied reverberation modeling. Anthony I. Eller and Kevin D. Heaney (OASIS, Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, ellera@oasislex.com)

The historical development of reverberation modeling is a story driven by both supporting and sometimes conflicting features of application requirements, measurement and computing capability, and scientific understanding. This paper presents an overview of how underwater reverberation modeling technology has responded to application needs and how this process has helped the community to identify and resolve related science issues. Emphasis is on the areas of System Design and Acquisition Support, Deployment and Operational Support, and Training Support. Gaps in our scientific knowledge are identified and recent advances are described that help push forward our collective understanding of how to predict and mitigate reverberation.

1:35

4pUW3. Reverberation models as an aid to interpret data and extract environmental information. Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com) and John R. Preston (Appl. Res. Lab., The Penn State Univ., State College, PA)

Reverberation measurements obtained with towed arrays are a valuable tool to extract information about the ocean environment. Preston pioneered the use of polar plots to display reverberation and superimpose the beam time series on bathymetry maps. As part of Rapid Environmental Assessment (REA) exercises Ellis and Preston [J. Marine Syst. 78, S359–S371, S372–S381] have used directional reverberation measurements to detect uncharted bottom features, and to extract environmental information using model-data comparisons. One enthusiast declared "This is like doing 100 simultaneous transmission loss runs and having the results available immediately." Though that was clearly an exaggeration and the results are not precise, the approach provides valuable information to direct more accurate and detailed surveys. The early work used range-independent (flat bottom) models for the model-data comparisons, while current work includes a range-dependent model based on adiabatic normal modes. A model has been developed which calculates reverberation from range-dependent bottom bathymetry, echoes from targets and discrete clutter objects, then outputs beam time series directly comparable with measured ones. Recent work has identified interesting effects in sea bottom sand dunes in the TREX experiments. This paper will provide an overview of the earlier work, and examples from the recent TREX experiment.

1:55

4pUW4. Reverberation data/model comparisons using transport theory. Eric I. Thorsos, Jie Yang, Frank S. Henyey, and W. T. Elam (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu)

Transport theory has been developed for modeling shallow water propagation and reverberation at mid frequencies (1–10 kHz) where forward scattering from a rough sea surface is taken into account in a computationally efficient manner. The method is based on a decomposition of the field in terms of unperturbed modes, and forward scattering at the sea surface leads to mode coupling that is treated with perturbation theory. Reverberation measurements made during TREX13 combined with extensive environmental measurements provide an important test of transport theory predictions. Modeling indicates that the measured reverberation was dominated by bottom reverberation, and the reverberation level in the 2−4 kHz band was observed to decrease as the sea surface conditions increased from a low sea state to a higher sea state. This suggests that surface forward scattering was responsible for the change in reverberation level. Results of data/model comparisons examining this effect will be shown. [Work supported by ONR Ocean Acoustics.]

Contributed Papers

2:15

4pUW5. Physics of backscattering in terms of mode coupling applied to measured seabed roughness spectra in shallow water. David P. Knobles (ARL, UT at Austin, PO BOX 8029, Austin, TX 78713-8029, dpknobles@yahoo.com)

Energy conserving coupled integral equations for the forward and backward propagating modal components have been previously developed [J. Acoust. Soc. Am. 130, 2673–2680 (2011)]. A rough seabed surface leads to a backscattered field and modifies the interference structure of the forward propagating field. Perturbation theory applied to the basic coupled integral equations allows for physical insight into the correlation of the roughness spectrum to the forward and backward modal intensities and cross mode coherence. This study applies the Nx2D integral equation coupled-mode approach to 3-D roughness measurements and examines the physics of the coupling of the forward and backward field components and computes the modal intensities as a function of azimuth. The roughness measurements were made in about 20 m of water off Panama City, Florida. [Work supported by ONR Code 322 OA.]

2:30

4pUW6. Energy conservation via coupled modes in waveguides with an impedance boundary condition. Steven A. Stotts and Robert A. Koch (Environ. Sci. Lab., Appl. Res. Labs/The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, stotts@arlut.utexas.edu)

A statement of energy conservation for a coupled mode formulation with real mode functions and eigenvalues has been demonstrated to be consistent with the statement of conservation derivable from the Helmholtz equation. The restriction to real mode functions and eigenvalues precludes coupled mode descriptions with waveguide absorption or untrapped modes. The demonstration, along with the derivation of the coupled mode range equation, relies on orthonormality in terms of a product of two modal depth functions integrated to infinite depth. This paper shows that energy conservation and the derivation of the coupled mode range equation can be extended to complex mode functions and eigenvalues, and that energy is conserved for ocean waveguides with a penetrable bottom boundary at a finite depth beneath any range dependence. For this, the penetrable bottom boundary is specified by an impedance condition for the mode functions. The new derivations rely on completeness and a modified orthonormality statement. Mode coupling is driven solely by waveguide range dependence. Thus, the form of the range equation and the values of the coupling coefficients are unaffected by a finite depth waveguide. Applications of energy conservation to examine the accuracy of a numerical coupled mode calculation are presented.

2:45

4pUW7. Effect of channel impulse response on matched filter performance in the 2013 Target and Reverberation Experiment. Mathieu E. Colin (Acoust. and Sonar, TNO, Postbus 96864, Den Haag 2509 JG, Netherlands, mathieu.colin@tno.nl), Michael A. Ainslie (Acoust. and Sonar, TNO, The Hague, Netherlands), Peter H. Dahl, David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Sean Pecknold (Underwater Surveillance and Communications, Defence Res. and Development Canada , Dartmouth, NS, Canada), and Robbert van Vossen (Acoust. and Sonar, TNO, The Hague, Netherlands)

Active sonar performance is determined by the characteristics of the target, the sonar system and the effect of the environment on the received waveform. The two main influences of the environment are propagation effects and the contamination of the target echo with a background. The ambient noise and reverberation are mitigated by means of signal processing, mostly through beamforming and matched-filtering. The improvement can be quantified by the signal to noise ratios before and after processing. Propagation effects can have a large influence on the gains obtained by the processing. To study the effect of the channel on the matched filter performance, broadband channel impulse responses were modeled and compared to measurements acquired during the Office of Naval Research-funded 2013 Target and Reverberation Experiment (TREX). In shallow water, a large time spread is often observed, reducing the effectiveness of the matched filter. TREX data show, however, a limited time spread. Model predictions indicate that this could be caused by a rough sea-surface, which while increasing propagation loss, at the same time increases matched filter gain.

3:00–3:15 Break

3:15

4pUW8. Using physical oceanography to improve transmission loss calculations in undersampled environments. Cristina Tollefsen and Sean Pecknold (Defence Res. and Development Canada, P. O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, cristina.tollefsen@gmail.com)

The vertical sound speed profile (SSP) is a critical input to any acoustic propagation model. However, even when measured SSPs are available they are frequently noisy "snapshots" of the SSP at a single moment in time and space and do not fully capture changes such as solar heating and wind-driven mixing that can significantly affect shallow water propagation on time scales of less than a day. Furthermore, SSPs measured in the field may not extend to the ocean bottom and are often based on measured profiles of temperature with an implicit assumption of constant salinity. In April–May 2013, the Target and Reverberation Experiment (TREX) was conducted in the Northeastern Gulf of Mexico near Panama City, Florida, a region strongly affected by local wind forcing, freshwater inputs, and the presence of a warm-core Gulf of Mexico Loop Current eddy ("Eddy Kraken") offshore of the experimental site. "Synthetic" SSPs were constructed for the trial area by combining knowledge of the physical oceanography and water masses in the area with the measured SSPs that were available. Transmission loss was modelled using both synthetic and measured SSPs and the results will be compared with measured transmission loss.

3:30

4pUW9. Analytic formulation for broadband rough surface and volumetric scattering including matched-filter range resolution. Wei Huang, Delin Wang, and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., 006 Hayden Hall, 370 Huntington Ave., Boston, MA 02115, huang.wei1@husky.neu.edu)

An analytic formulation is derived for the broadband scattered field from a randomly rough surface based on Green's theorem employing perturbation theory. The matched filter is applied to resolve the scattered field within the range resolution footprint of a broadband imaging system. Statistical moments of the scattered field are then expressed in terms of the second moment characterization of the scattering surface. The broadband diffuse reverberation depends on the rough surface spectrum evaluated over a range of wavenumbers, centered at the Bragg wavenumber corresponding to the center frequency of the broadband pulse and extending to wavenumbers proportional to the signal bandwidth. A corresponding analytic broadband volume scattering model is derived from the Rayleigh-Born approximation to Green's theorem.

3:45

4pUW10. Objective identification of the dominant seabed scattering mechanism. Gavin Steininger (SEOS, U Vic, 201 1026 Johnson St., Victoria, BC v7v 3n7, Canada, gavin.amw.steininger@gmail.com), Charles W. Holland (SEOS, U Vic, State College, Pennsylvania), Stan E. Dosso, and Jan Dettmer (SEOS, U Vic, Victoria, BC, Canada)

This paper develops and applies a quantitative inversion procedure for scattering-strength data to determine the dominant scattering mechanism (surface and/or volume scattering) and to estimate the relevant scattering parameters and their uncertainties. The classification system is based on trans-dimensional Bayesian inversion with the deviance information criterion used to select the dominant scattering mechanism. Scattering is modeled using first-order perturbation theory as due to one of three mechanisms: interface scattering from a rough seafloor, volume scattering from a heterogeneous sediment layer, or mixed scattering combining both interface and volume scattering. The classification system is applied to six simulated test cases where it correctly identifies the true dominant scattering mechanism as having greater support from the data in five cases; the remaining case is indecisive. The approach is also applied to measured backscatter-strength data from the Malta Plateau where volume scattering is determined as the dominant scattering mechanism. This conclusion and the scattering/geoacoustic parameters estimated in the inversion are consistent with properties from previous inversions and/or with core measurements from the site. In particular, the scattering parameters are converted from the continuous scattering models used in the inversion to the equivalent discrete scattering parameters, which are found to be consistent with properties of the cores. [Work supported by ONR.]

4:00

4pUW11. Laboratory measurements of backscattering strengths from two-types of artificially roughened sandy bottoms. Su-Uk Son (Dept. of Marine Sci. and Convergent Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 426-791, South Korea, suuk2@hanyang.ac.kr), Sungho Cho (Maritime Security Res. Ctr., Korea Inst. of Ocean Sci. & Technol., Ansan, Gyeonggi-do, South Korea), and Jee Woong Choi (Dept. of Marine Sci. and Convergent Technol., Hanyang Univ., Ansan, Gyenggi-do, South Korea)

In the case of sandy bottom, the backscattering from the interface roughness is significantly dominant compared to that from the volume inhomogeneities, and the power spectrum of interface roughness thus becomes the most important factor to control the scattering mechanism. Backscattering strength measurements with a 50-kHz signal were made for two types of roughness (smooth and rough interfaces) which were artificially formed on a 0.5-m thick sandy bottom in a 5-m deep water tank. The roughness profiles were estimated by the arrival time analysis of 5-MHz backscattering signals emitted by the transducer moving parallel to the interface at a speed of 1 cm/s, which were then Fourier transformed to yield power spectra. In this talk, the measurements of backscattering strength as a function of grazing angle in a range of 35 to 90° are presented. Finally, the effect of different roughness types on the scattering strength will be discussed in comparison with the predictions obtained by theoretical scattering model including the perturbation and Kirchhoff approximations. [This research was supported by the Agency for Defense Development, Korea.]

4:15

4pUW12. Multistatic performance prediction for Doppler-sensitive waveforms in a shallow-water environment. Cristina Tollefsen (Defence Res. and Development Canada, P. O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, cristina.tollefsen@gmail.com)

Navies worldwide are now operationally capable of exploiting multistatic sonar technology. One of the purported advantages of multistatics when detecting directional targets should be the increased probability of receiving a strong reflection at one of the multistatic receivers. However, it is not yet clear (or intuitive) how best to deploy multistatic-capable assets to achieve particular mission objectives. The Performance Assessment for Tactical Systems (PATS) software was recently developed by Maritime Way Scientific under contract to Defence Research and Development Canada as a research tool to assist in exploring different approaches to multistatic performance modelling. Beginning with a user-defined environment and sensor layout, PATS uses transmission loss and reverberation model results to calculate signal excess at each grid point in the model domain. Monte Carlo simulations using many realizations of target tracks allow for the calculation of the cumulative probability of detection as a means to assess performance. Results will be presented comparing the shallow-water performance of monostatic and multistatic sensors using frequency-modulated and Doppler-sensitive waveforms as well as omnidirectional and directional targets in a variety of realistic military scenarios.

4:30

4pUW13. Twinkling exponents for backscattering by spheres in the vicinity of airy caustics associated with reflections by corrugated surfaces. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

High frequency sound reflected by corrugated surfaces produce caustic networks relevant to sea surface reflection [Williams *et al.*, J. Acoust. Soc. Am. 96, 1687–1702 (1994)]. When a sphere is positioned sufficiently far from the reflecting surface, it may be close to an Airy caustic which causes a significant increase in the backscattering [Dzikowicz and Marston, J. Acoust. Soc. Am. 116, 2751–2758 (2004)] for signals that bounce only once off of the focusing surface. For simplicity, here, it is assumed that those signals may be distinguished from the earlier direct echo from the sphere and the later (and sometimes stronger) doubly focused echo from the sphere [Dzikowicz and Marston, J. Acoust. Soc. Am. 118, 2811–2819 (2005)]. In 1977, M. V. Berry noticed that the third and higher intensity moments of wavefields containing caustics can increase in proportion to k^{ν}, where k is the wavenumber and ν is a "twinkling exponent" determined by the

dependencies of the intensity and focal volume on k. Assuming that the sphere is impenetrable and sufficiently large that its direct scattering depends only weakly on k, for the single-bounce backscattering by a sphere considered here (the easiest situation for applying Berry's analysis) the predicted exponent for the third moment is $\nu = 1/3$.

THURSDAY EVENING, 30 OCTOBER 2014 — 7:30 P.M. TO 9:00 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. On Tuesday the meetings will begin at 8:00 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:30 p.m. On Wednesday evening, the Technical Committee on Biomedical Acoustics will meet starting at 7:30 p.m. On Thursday evening, the meetings will begin at 7:30 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:

Animal Bioacoustics	Lincoln
Musical Acoustics	Santa Fe
Noise	Marriott 3/4
Psychological and Physiological Acoustics	Marriott 1/2
Signal Processing in Acoustics	Indiana G
Underwater Acoustics	Indiana F

Session 5aBA

Biomedical Acoustics: Cavitation Control and Detection Techniques

Kevin J. Haworth, Cochair
Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

Oliver D. Kripfgans, Cochair
Dept. of Radiology, Univ. of Michigan, Ann Arbor, MI 48109-5667

Invited Papers

8:00

5aBA1. Detection and control of cavitation during blood–brain barrier opening: Applications and clinical considerations. Meaghan A. O'Reilly, Ryan M. Jones, Alison Burgess, Cassandra Tyson, and Kullervo Hynynen (Physical Sci., Sunnybrook Res. Inst., 2075 Bayview Ave., Rm. C713, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

Microbubble-mediated opening of the blood–brain barrier (BBB) using ultrasound is a targeted technique that provides a transient time window during which circulating therapeutics that are normally restricted to the vasculature can pass into the brain. This effect has been associated with increases in cavitation activity of the circulating microbubbles, and our group has previously described a method to actively control treatments in pre-clinical rodent models based on acoustic emissions recorded by a single transducer. Recently, we have developed a clinical-scale receiver array capable of detecting bubble activity through *ex vivo* human skullcaps starting at pressure levels below the threshold for BBB opening. The use of this array to spatially map cavitation activity in the brain during ultrasound therapy will be discussed, including considerations for compensating for the distorting effects of the skull bone. Additionally, results from pre-clinical investigations examining safety and therapeutic potential will be presented, and receiver design considerations for both pre-clinical and clinical scale systems will be discussed.

8:20

5aBA2. Passive acoustic mapping of stable and inertial cavitation during ultrasound therapy. Christian Coviello (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), James Choi (Dept. of BioEng., Imperial College, London, United Kingdom), Jamie Collin, Robert Carlisle (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Miklos Gyongy (Faculty of Information Technol. and Bionics, Pazmany Peter Catholic Univ., Prague, Hungary), and Constantin C. Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Oxford, Oxfordshire OX3 7DQ, United Kingdom, constantin.coussios@eng.ox.ac.uk)

Accurate spatio-temporal characterization, quantification, and control of the type and extent of cavitation activity is crucial for a wide range of therapeutic ultrasound applications, ranging from ablation to sonothrombolysis, opening of the blood-brain barrier and drug delivery for cancer. Passive Acoustic Mapping (PAM) is a technique that utilizes arrays of acoustic detectors, typically coaxially aligned or coincident with the therapeutic elements, to receive acoustic emissions outside the main frequency band of the therapy pulse. The signals received by each detector are then filtered in the frequency domain into harmonics and ultra/subharmonics of the fundamental therapeutic frequency and other broadband components, and subsequently beamformed using a multi-correlation algorithm, which uses measures of similarity between the signals rather than time-of-flight information in order to map sources of non-linear emissions in real time. 2D and 3D cavitation maps obtained using time exposure acoustics beamforming will be presented, and juxtaposed to the greater spatial resolution but increased computational complexity afforded by more advanced algorithms such as the Robust Capon Beamformer (RCB). The spatial correlation between cavitation maps produced using PAM and the associated therapeutic effect will also be discussed in the context of cavitaion-enhanced ablation and drug delivery.

8:40

5aBA3. Image-guided sonothrombolysis in a stroke model with a cavitation delivery and monitoring system. Francois Vignon, William T. Shi (Ultrasound Imaging and Therapy, Philips Res. North America, 345 Scarborough Rd., Briarcliff Manor, NY 10510, francois.vignon@philips.com), Jeffry Powers (Philips Ultrasound, Bothell, WA), Feng Xie, Juefei Wu, Shunji Gao, John Lof, and Thomas R. Porter (Cardiology, Univ. of NE Medical Ctr., Omaha, NE)

Microbubbles (MB) and ultrasound (US) can dissolve intra-arterial thrombi. In order to reproducibly deliver the correct cavitation dose and ensure treatment efficacy and safety, we designed a therapeutic US mode with cavitation monitoring. Therapy delivery and recording of the MB signal are achieved with a sector imaging probe. Monitoring is achieved by spectrally analyzing the MB signal: ultraharmonics are a marker of stable cavitation (SC) and broadband noise characterizes inertial cavitation (IC). We used the system in a pig model. Thrombotic occlusions were created by injecting 4-hour old clots bilaterally into the internal carotids. Forty pigs were randomized to either 2.4 MI, 5 μs pulses with MBs; 1.7 MI, 20 μs pulses with MBs; and 2.4 MI, 5 μs pulses without MBs. Angiographic recanalization rates were compared. Cavitation as a function of MI was estimated *in vivo*. Dominant SC started at an applied MI of 0.6

(0.3MI *in situ* after derating by skull attenuation). Dominant IC was estimated to start at an applied MI of 0.9 (0.6 *in situ*). Thus, all therapy settings were in the IC regime. The 2.4MI + MB setting was the most effective (100% recanalization) vs 38% for the 1.7MI + MB and 50% for 2.4 MI without MBs (both $p<0.05$ compared to 2.4MI + MB). No signs of hemorrhage were found in any animal. In conclusion, higher IC levels are most effective for thrombus dissolution. Spectral analysis techniques can be used to plan and monitor the therapy.

9:00

5aBA4. Timing of high intensity pulses for myocardial cavitation-enabled therapy. Douglas Miller, Chunyan Dou (Radiology, Univ. of Michigan, 3240A Medical Sci. I, 1301 Catherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), Gabe E. Owens (Pediatrics, Univ. of Michigan, Ann Arbor, MI), and Oliver Kripfgans (Radiology, Univ. of Michigan, Ann Arbor, MI)

Ultrasound pulses intermittently triggered from an ECG signal can interact with circulating microbubbles to produce myocardial cavitation microlesions, which may enable tissue-reduction therapy. The timing of therapy pulses relative to the ECG was investigated to identify the optimal trigger point with regard to physiological response and microlesion production. Rats were anesthetized, prepared for ultrasound, placed in a heated water bath, and treated with 1.5 MHz focused ultrasound pulses aimed by 8 MHz imaging. Initially, rats were treated for 1 min with triggering at each of six different points in the ECG while monitoring blood pressure. Premature complexes, a useful indicator of efficacy, were seen in the ECG, except during early systole. Premature complexes corresponded with blood pressure pulses for triggering during diastole, but not during systole. Next, triggering at three of the time points, end diastole, end systole, or mid-diastole, was tested for the impact on microlesion creation. Microlesions stained by Evans blue dye were scored in frozen sections. There was no statistically significant variation in cardiomyocyte injury. The end of systole was identified as an optimal trigger time point which yielded ECG complexes and substantial cardiomyocyte injury, but minimal cardiac functional disruption.

9:20

5aBA5. Cavitation threshold determination —Can we do it? Gail ter Haar, John Civale, Ian Rivens, and Marcia Costa (Phys., Inst. of Cancer Res., Phys. Dept., Royal Marsden Hospital, Sutton, Surrey SM2 5PT, United Kingdom, gail.terhaar@icr.ac.uk)

As clinical applications, which harness acoustic cavitation, become more commonplace, it becomes more and more important to be able to determine the threshold pressures at which it is likely to occur. In our studies, we have used a suite of different detection techniques in an effort to determine these thresholds. These include passive cavitation detection, transducer impedance monitoring, and visual appearance. Different methods of acoustic signal processing have been compared. The resultant cavitation thresholds will be discussed.

9:40

5aBA6. Monitoring boiling histotripsy with bubble-based ultrasound techniques. Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washigton, 1013 NE 40th St., Seattle, WA 98105, va.khokhlova@gmail.com), Michael Canney (INSERM U556, Lyon, France), Julianna Simon, Tatiana Khokhlova, Joo-Ha Hwang, Adam Maxwell, Michael Bailey, Oleg Sapozhnikov, Wayne Kreider, and Lawrence Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washigton, Seattle, WA)

Cavitation phenomena have been always considered as a predominant mechanism of concern in mechanical tissue damage induced by therapeutic ultrasound. Corresponding methods have been developed to monitor cavitation. Recently, a new high intensity focused ultrasound technology, called boiling histotripsy (BH), was introduced, in which the major physical phenomenon that initiates mechanical tissue damage is vapor bubble growth associated with rapid tissue heating to boiling temperatures. Caused by nonlinear propagation effects and the development of high-amplitude shocks, this tissue heating is localized in space and can lead to boiling within milliseconds. Once a boiling bubble is created, interaction of shock waves with the cavity results in tissue disintegration. While the incident shocks can lead to cavitation phenomena and accompanying broadband emissions, the presence of a millimeter-sized vapor cavity in tissue produces strong echogenicity in ultrasound (US) imaging that can be exploited with B-mode diagnostic ultrasound. Various other methods of imaging boiling histotripsy, including passive cavitation detection (PCD), Doppler or nonlinear pulse-inversion techniques, and high speed photography in transparent gel phantoms are also overviewed. The role of shock amplitude as a metric for mechanical tissue damage is discussed. [Work supported by NIH EB007643, T32DK007779, and NSBRI through NASA NCC 9-58.]

10:00

5aBA7. Control of cavitation through coalescence of cavitation nuclei. Timothy L. Hall, Alex Duryea, and Hedieh Tamaddoni (Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109, hallt@umich.edu)

Therapeutic ultrasound in the form of SWL, HIFU, or histotripsy frequently generates cavitation nuclei (bubbles 1–10 um radius), which can persist up to about 1 s before dissolving. These nuclei can attenuate and reflect propagation of acoustic fields reducing SWL efficiency, enhancing HIFU heating, or shifting the location of a histotripsy focal zone making procedures less predictable. Depending on their location, nuclei can also directly cause tissue damage when a high amplitude sound field causes them to undergo inertial cavitation. These undesirable effects can be reduced by using a low amplitude sound field (MI <1) to stimulate coalescence of nuclei through primary and secondary Bjerknes forces. We will show nuclei coalescence significantly reduces sound field attenuation, improves SWL breakup of model kidney stones, and reduces collateral damage in soft tissues. We also show techniques for designing the non-focal acoustic fields for efficient coalescence with 3D printed acoustic lenses. Timothy Hall has a consulting arrangement with Histosonics, Inc., which has licensed intellectual property related to this abstract.

10:20–10:30 Break

Contributed Papers

10:30

5aBA8. Ultraharmonic intravascular ultrasound imaging with commercial 40 MHz catheter: A feasibility study. Himanshu Shekhar, Ivy Awuor, Steven Huntzicker, and Marvin M. Doyley (Univ. of Rochester, 345 Hopeman Bldg., University of Rochester River Campus, Rochester, NY 14627, himanshuwaits@gmail.com)

The abnormal growth of the vasa vasorum is characteristic of life-threatening atherosclerotic plaques. Intravascular ultraharmonic imaging is an emerging technique that could visualize the vasa vasorum and help clinicians identify life-threatening plaques. Implementing this technique on commercial intravascular ultrasound (IVUS) systems could to accelerate its clinical translation. Our previous work has demonstrated ultraharmonic IVUS imaging with a modified clinical system that was equipped with a commercial 15 MHz peripheral imaging catheter. In the present study, we investigated the feasibility of ultraharmonic imaging with a commercially available 40 MHz coronary imaging catheter. We imaged a flow phantom that had contrast agent microbubbles (Targestar-P-HF, Targeson Inc., CA) perfused in side channels parallel to its main lumen. The transducer was excited at 30 MHz using 10% bandwidth chirp-coded pulses. The ultraharmonic response at 45 MHz was isolated and preferentially visualized using pulse inversion and digital filtering. Side channels with 900 μm and 500 μm diameter were detected with contrast-to-tissue ratios approaching 10 dB for clinically relevant microbubble concentrations. The results of this study indicate that ultraharmonic imaging is feasible with commercially available coronary IVUS catheters, which may facilitate its widespread application in preclinical research and clinical imaging.

10:45

5aBA9. A method to calibrate the absolute receive sensitivity of spherically focused, single-element transducers. Kyle T. Rich and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Quantitative acoustic measurements of microbubble behavior, including scattering and emissions from cavitation, would be facilitated by improved calibration of transducers making absolute pressure measurements. In particular, appropriate methods are needed for wideband calibration of focused passive cavitation detectors. Here, a substitution method was developed to characterize the absolute receive sensitivity of two spherically focused, single-element transducers (center transmit frequencies 4 and 10 MHz). Receive calibrations were obtained by transmitting and receiving a broadband pulse between the two focused transducers in a pitch-catch, confocally aligned configuration, separated by a distance equal to the sum of the two focal lengths. A calibrated hydrophone was substituted to measure the pressure field in the plane of each receiver's surface. The frequency dependent receive sensitivity at the focus was then calculated for each transducer as the ratio of the receiver-measured voltage and the average hydrophone-measured pressure amplitude across the receiver surface. Calibrations were validated by generating an approximately spherically spreading, broadband pressure wave at the focus of each transducer using a 2-mm diameter transducer and comparing the absolute acoustic pressure measured by each focused transducer to that measured by a calibrated hydrophone.

11:00

5aBA10. Instigation and monitoring of inertial cavitation from nanoscale particles using a diagnostic imaging platform and passive acoustic mapping. Christian Coviello, James Kwan, Susan Graham, Rachel Myers, Apurva Shah, Penny Probert Smith, Robert Carlisle, and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, ORCRB, Oxford OX3 7DQ, United Kingdom, christian.coviello@eng.ox.ac.uk)

Inertial cavitation nucleated by microbubble contrast agents has been recently shown to enhance extravasation and improve the distribution of anti-cancer agents during ultrasound (US)-enhanced delivery. However, microbubbles require frequent replenishment due to their rapid clearance and destruction upon US exposure and are unable to extravasate into tumor tissue due to their large size. A new generation of gas-stabilizing polymeric cup-shaped nanoparticles, or "nanocups" (NCs), have been formulated to a size that enables exploitation of the enhanced permeability and retention effect for intratumoral accumulation. NCs provide sustained inertial cavitation, as characterized by broadband emissions, at peak rarefactional pressures readily achievable by diagnostic ultrasound systems. This enables the use of a single low-cost system for B-mode treatment guidance, instigation of cavitation, and real-time passive acoustic mapping (PAM) of the location and extent of cavitation activity during therapy. The significant lowering of the inertial cavitation threshold in the presence of NCs as characterized by PAM is first quantified *in-vitro*. *In-vivo* and *ex-vivo* results in xenograft-implanted tumor bearing mice further evidence the strong presence of inertial cavitation detectable in the tumor at diagnostic levels of US intensity, as confirmed by PAM images overlaid on B-mode in real-time.

11:15

5aBA11. Passive cavitation imaging with nucleic acid-loaded microbubbles in mouse tumors. Man M. Nguyen, Jonathan A. Kopechek, Bima Hasjim, Flordeliza S. Villanueva, and Kang Kim (Dept. of Medicine, Univ. of Pittsburgh, 3550 Terrace St., 562 Scaife Hall, Pittsburgh, PA 15261, manmnguyen@gmail.com)

Ultrasound-targeted microbubble (MB) destruction has been used to deliver nucleic acids to cancer cells for therapeutic effect. Identifying both the location and cavitation activities of the MBs is needed for efficient and effective treatment. In this study, we implemented passive cavitation imaging into a commercially available ultrasound open platform (Verasonics) for a 128-element linear array transducer, centered at 5 MHz, and applied it to an *in-vivo* mouse tumor model. Cationic lipid MBs were loaded with a transcription factor decoy that suppresses STAT3 signaling and inhibits tumor growth in murine squamous cell carcinomas. During systemic MB infusion, ultrasound pulses (4 or 20 cycles) were delivered with a 1-MHz single-element transducer (0.4–1.4MPa peak pressures). Channel data were offline beamformed, band-pass filtered, subtracted from reference images acquired without MBs, and co-registered with B-mode images. During MB infusion, harmonics and broadband emissions were detected in the tumor with both frequency spectra and cavitation images. For 4-cycle 0.4 MPa pulses, harmonic signals at 5 MHz and broadband signals 3–7 MHz were 23 dB and at least 5 dB greater with MBs than without MBs, respectively. These preliminary results demonstrate the feasibility of *in-vivo* passive cavitation imaging and could lead to further studies for optimizing US/MB-mediated delivery of nucleic acids to tumors.

11:30

5aBA12. Non-focal acoustic lens designs for cavitation bubble consolidation. Hedieh A. Tamaddoni, Alexander Duryea, and Timothy L. Hall (Univ. of Michigan, 2740 Barclay Way, Ann Arbor, MI 48105, alavi@umich.edu)

During shockwave lithotripsy, cavitation bubbles form on the surface of urinary stones aiding in the fragmentation process. However, shockwaves can also produce pre-focal bubbles, which may shield or block subsequent shockwaves and potentially induce collateral tissue damage. We have previously shown *in-vitro* that low amplitude acoustic waves can be applied to actively stimulate bubble coalescence and help alleviate this effect. A traditional elliptical transducer lens design produces the maximum focal gain possible for a given aperture. From experiments and simulation, we have found that this design is not optimal for bubble consolidation as the primary and secondary Bjerknes forces may act against each other and the effective field volume is too small. For this work, we designed and constructed non-focal transducer lenses with complex surface geometries using rapid prototyping stereolithography to produce more effective acoustic fields for bubble consolidation during lithotripsy or ultrasound therapy. We demonstrate a design methodology using an inverse problem technique to map the desired acoustic field back to the surface of the transducer lens to determine the correct phase shift at every point on the lens surface. This method could be applied to other acoustics problems where non-focused acoustic fields are desired.

11:45

5aBA13. Scavenging dissolved oxygen via acoustic droplet vaporization. Kirthi Radhakrishnan, Christy K. Holland, and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cardiovascular Ctr. 3972, 231 Albert Sabin Way, Cincinnati, OH 45267, radhakki@ucmail.uc.edu)

Acoustic droplet vaporization (ADV) has been investigated for capillary hemostasis, thermal ablation, and ultrasound imaging. The maximum diameter of a microbubble produced by ADV depends on the gas saturation of the surrounding fluid. This dependence is due to diffusion of dissolved gases from the fluid into the perfluoropentane (PFP) microbubble. This study investigated the change in oxygen concentration in the surrounding fluid after ADV. Albumin-shelled PFP droplets in air-saturated saline (1:30, v/v) were continuously pumped through a flow system and insonified by a focused 2-MHz single-element transducer to induce ADV. B-mode image echogenicity was used to determine the ADV threshold pressure amplitude. The dissolved oxygen concentration in the fluid upstream and downstream of the insonation region was measured using inline sensors. Droplet size distributions were measured before and after ultrasound exposure to determine the ADV transition efficiency. The ADV pressure threshold at 2 MHz was 1.7 MPa (peak negative). Exposure of PFP droplets to ultrasound at 5 MPa peak negative pressure caused the dissolved oxygen content in the surrounding fluid to decrease from 88 $\pm$ 3% to 20 $\pm$ 4%. The implications of oxygen scavenging during ADV will be discussed.

12:00

5aBA14. Effects of rose bengal on cavitation cloud behavior in optically transparent gel phantom investigated by high-speed observation. Jun Yasuda, Takuya Miyashita (Dept. of Commun. Eng., Tohoku Univ., 6-6-06-5 Aramakiazaaoba, Aoba, Sendai, Miyagiken 980-0801, Japan, j_yasuda@ecei.tohoku.ac.jp), Kei Taguchi (Dept. of Biomedical Eng., Tohoku Univ., Sendai, Japan), Shin Yoshizawa (Dept. of Commun. Eng., Tohoku Univ., Sendai, Japan), and Shin-ichiro Umemura (Dept. of Biomedical Eng., Tohoku Univ., Sendai, Japan)

Sonodynamic treatment is a non-thermal ultrasonic method using sonochemical effect of cavitation bubbles. Rose bengal (RB) is sonochemically active and reduces cavitation threshold and therefore has potential to be an agent for sonodynamic treatment. For the effectiveness and safety of the treatment, controlling cavitation is crucial. In our previous study, we have suggested high-intensity focused ultrasound (HIFU) employing second-harmonic superimposition, which can control cavitation cloud generation by superimposing the second harmonic onto the fundamental. In this study, to investigate the effects of RB on cavitation behavior, a polyacrylamide gel phantom containing RB was exposed to second-harmonic superimposed ultrasound and the generated cavitation bubbles were observed by a high-speed camera. The gel contained three different concentrations of RB, 0, 1, and 10 mg/L. The ultrasonic intensity and exposure duration were 40 kW/cm^2 and 100 μs, respectively. The fundamental frequency was 0.8 MHz. In the results, the amount of the incepted cloud became higher and the lifetime of bubbles became longer as the RB concentration increased at high reproducibility. The observed RB concentration dependence suggests that the amount of cavitation bubbles can be controlled using second-harmonic superimposition. The observed lifetime extension of bubbles can not only promote sonochemical but also enhance thermal bioeffect.

12:15–12:30 Panel Discussion

FRIDAY MORNING, 31 OCTOBER 2014 — INDIANA E, 10:00 A.M. TO 1:00 P.M.

Session 5aED

Education in Acoustics: Hands-On Acoustics Demonstrations for Indianapolis Area Students

Uwe J. Hansen, Cochair
Chemistry& Physics, Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

Andrew C. H. Morrison, Cochair
Natural Science Department, Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomenon. In this session "Hands-On" demonstrations will be set-up for a group of middle school students from the Indianapolis area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students' educational development and is part of the larger "Listen Up" education outreach effort by the ASA.

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 email Uwe Hansen (uhansen@indstate.edu) or Andrew C. H. Morrison (amorriso@jjc.edu).

Session 5aNS

Noise: Transportation Noise, Soundscapes, and Related Topics

Alan T. Wall, Chair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Chair's Introduction—9:45

Contributed Papers

9:50

5aNS1. Traffic monitoring with noise: Investigations on an urban seismic network. Nima Riahi and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, nriahi@ucsd.edu)

Traffic in urban areas generates not only acoustic noise but also much seismic noise. The latter is typically not perceptible by humans but could, in fact, offer an interesting data source for traffic information systems. To explore the potential for this, we study a 5300 geophone network, which covered an area of over 70 km^2 in Long Beach, CA, and was deployed as part of a hydrocarbon industry survey. The sensors have a typical spacing of about 100 m, which presents a two-sided processing challenge here: signals beyond a few receiver spacings from the sources are often strongly attenuated and scattered whereas nearby receiver signals may contain complicated near-field effects. We illustrate how we address this issue and give three simple applications: counting cars on a highway section, classifying different types of vehicles passing along a road, and measuring time and take-off velocity of aircraft at Long Beach airport. We discuss future work toward traffic monitoring and also possible connections with acoustical problems.

10:05

5aNS2. Impact of AMX-A1 military aircraft operations on the acoustical environment close to a Brazilian airbase. Olmiro C. de Souza and Stephan Paul (UFSM, Acampamento, 569, Santa Maria, Santa Maria 97050003, Brazil, olmirocz.eac@gmail.com)

Military aircraft operating on airbases usually have a considerable impact on the neighborhood. While the impact of civil aircraft operations can be modeled by commercially available software, the same is hardly possible for military aircraft as no EPNL data are available for such aircraft and flight path are not restricted to those used by civilian operations. Therefore, in this work, the noise impact of AMX-A1 aircraft operating at a Brazilian airbase was evaluated from measurements originally intended for calibration of the noise map of an university that is in the vicinity. From the data, it was possible to obtain $L_{Aeq10min}$ with and without jet noise to see how much does AMX operations influence in the total measurement. Sound exposure levels (SEL) were also calculated. It was found that depending of AMX procedure (Approach, Departure, Touch, Go, etc.), jet noise increases the $L_{Aeq10min}$ up to 10 dB and SEL values reaches 96, 6 dBA in sensitive areas. It will be discussed if A-weighted sound power level can be estimated form the data considering the aircraft as a point source in free field.

10:20

5aNS3. The effect of long-range propagation on contra-rotating open rotor en-route noise levels. Upol Islam (Inst. of sound and Vib. Res. (ISVR), Univ. of Southampton, Highfield Campus, Bldg. 13, Rm. 2009, Southampton, Hampshire SO17 1BJ, United Kingdom, ui1d11@soton.ac.uk)

The purpose was to calculate the en-route noise level produced by an advanced contra-rotating open rotor (CROR) powered aircraft. The en-route noise is defined as the noise produced by an aircraft in high altitude operation (>3000 m) measured by a microphone 1.2 m above ground level. Calculations were performed for three different aircraft operating conditions—cruise, climb, and descent. For each calculation, the aircraft noise source was modeled as an isolated CROR engine. This noise model was determined from experimental measurements made in a transonic wind tunnel using a 1/6th-scale open rotor rig. En-route noise levels were calculated using the whole aircraft noise prediction code SOPRANO. The CROR noise model were input into SOPRANO and long-distance propagation was calculated using the ray-tracing code APHRODITE, which is implemented within SOPRANO. This ray tracing code requires atmospheric wind speed, wind direction, temperature, and humidity profiles, which were collected from historical data around Europe. The ray tracing method divides the atmosphere up into a number of layers. Meteorological parameters were assumed to vary linearly between the values specified at the boundaries of each layer. Numerous simulations are conducted using different atmospheres in order to assess the impact of atmospheric conditions on the en-route noise levels.

10:35

5aNS4. Gaps in the literature on the effects of aircraft noise on children's cognitive performance. Matthew Kamrath and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, kamrath64@gmail.com)

In the past two decades, several major studies have indicated that chronic aircraft noise exposure negatively impacts children's cognitive performance. For example, the longitudinal Munich airport study (Hygge, Am. Psychol. Soc., 2002) demonstrated that noise adversely affects reading ability, memory, attention, and speech perception. Moreover, the cross-sectional RANCH study (Stansfeld, Lancet, 2005) found a linear correlation between extended noise exposure and reduced reading comprehension and recognition memory. This presentation summarizes these and other recent studies and discusses four key areas in need of further research (ENNAH Final Report Project No. 226442, 2013). First, future studies should account for all of the following confounding factors: socioeconomic variables, daytime and nighttime aircraft, road, and train noise, and air pollution. Second, multiple noise metrics should be evaluated to determine if the character of the noise alters the relationship between noise and cognition. Third, future research should explore the mitigating effects of improved classroom acoustics and exterior sound insulation. Finally, additional longitudinal studies are necessary: (1) to establish a causal relationship between aircraft noise and cognition; and (2) to understand how changes in the duration of the exposure and in the age of the students influence the relationship. [Work supported by FAA PARTNER Center of Excellence.]

10:50

5aNS5. Acoustic absorption of green roof samples commercially available in southern Brazil. Ricardo Brum, Stephan Paul (Centro de Tecnologia, Universidade Federal de Santa Maria, Rua Erly de Almeida Lima, 650, Santa Maria, RS 97105-120, Brazil, ricardozbrum@yahoo.com.br), Andrey R. da Silva (Centro de Engenharias da Mobilidade, Universidade Federal de Santa Catarina, Joinville, Brazil), and Tenile Piovesan (Centro de Tecnologia, Universidade Federal de Santa Maria, Santa Maria, RS, Brazil)

Previous investigations have shown that green roofs provide many environmental benefits, such as thermal conditioning, air cleaning, and rain water absorption. Nevertheless, information regarding acoustic properties, such as sound absorption and transmission loss is still sparse. This work presents measurements of the sound absorption coefficient of two types of green roofs commercially available in Brazil: the alveolar and the hexa system. Measurements were made in a reverberant chamber according to ISO-354 for different variations of both systems: the alveolar system with 2.5 cm of substrate with and without grass and 4 cm of substrate only. The hexa system was measured with layers of 4 and 6 cm of substrate without vegetation and 6 cm of substrate with a layer of vegetation of the sedum type. For all systems, high absorption coefficients were found for medium and high frequency limits ($\alpha \approx 0.7$) and low absorption in low frequencies ($\alpha \approx 0.2$). This was expected due to the highly porous structure of the substrate. The results suggest that the types of green roofs evaluated in this work could be a good approach to noise control in urban areas.

11:05

5aNS6. The perceived annoyance of urban soundscapes. Adam Craig, Don Knox, and David Moore (School of Eng. and Built Environment, Glasgow Caledonian Univ., 70 Cowcaddens Rd., Glasgow G4 0BA, United Kingdom, Adam.Craig@gcu.ac.uk)

Annoyance is one of the main factors that contribute to a negative view of environmental noise, and can lead to stress-related health conditions. Subjective perception of environmental sounds is dependent upon a variety of factors related to the sound, the geographical location, and the listener. Noise maps used to communicate information to the public about environmental noise in a given geographic location are based on simple noise level measurements and do not include any information regarding how perceptually annoying or otherwise the noise might be. This study involved subjective assessment by a large panel of listeners ($N = 200$) of a corpus of 60 pre-recorded urban soundscapes collected from a variety of locations around Glasgow City Centre. Binaural recordings were taken at three points during each 24 hour period in order to capture urban noise during day, evening, and night. Perceived annoyance was measured using Likert and numerical scales and each soundscape measured in terms of arousal and positive/negative valence. The results shed light on the subjective annoyance of environmental sound in a range of urban locations around Glasgow, and form the basis for development of environmental noise maps, which more fully communicate the effects of environmental noise to the public.

11:20

5aNS7. What comprises a healthy soundscape for the captive Southern White Rhinoceros *(Ceratotherium simum simum)*? Suzi Wiseman (Environ. Geography, Texas State Univ.-San Marcos, 3901 North 30th St., Waco, TX 76708, sw1210txstate@gmail.com), Preston S. Wilson (Mech. Eng., Univ. Texas at Austin, Austin, TX), and Frank Sepulveda (Geophys., Baylor Univ., Killeen, TX)

Many creatures, including the myopic rhinoceros, depend upon hearing and smell to determine their environment. Nature is dominated by meaningful biophonic and geophonic information quickly absorbed by soil and vegetation, while anthrophonic urban soundscapes exhibit vastly different physical and semantic characteristics, sound repeatedly reflecting off hard geometric surfaces, distorting and reverberating, and becoming noise. Noise damages humans physiologically, including reproductively, and likely damages other mammals. Rhinos vocalize sonically and infrasonically, but audiograms are unavailable. They generally breed poorly in urban zoos, where infrasonic noise can be chronic. Biological and social factors are studied, but little attention if any is paid to soundscape. We present a methodology to analyze the soundscapes of captive animals according to their hearing range. Sound metrics determined from recordings at various institutions can then be compared and correlations with the health and wellbeing of their animals can be sought. To develop this methodology we studied the sonic, infrasonic, and seismic soundscape experienced by the white rhinos at Fossil Rim Wildlife Center, one of the few U.S. facilities to successfully breed this species in recent years. Future analysis can seek particular parameters known to be injurious to human mammals, plus parameters known to invoke response in animals.

11:35

5aNS8. Shape optimization of acoustic horns using few design variables. Nilson Barbieri (Mech. Eng., PUCPR, Rua Imaculada Conceição, 1155, Curitiba, Paraná 80215-901, Brazil, nilson.barbieri@pucpr.br), Renato Barbieri (Mech. Eng., UDESC, Joinville, Santa Catarina, Brazil), Clebe T. Vitorino, and Key F. Lima (Mech. Eng., PUCPR, Curitiba, Brazil)

The main steps for design of the optimal geometry of acoustic horns employing numerical methods are: the definition of the domain and the restrictions and control of the boundary, the definition of the objective function and the frequency range of interest, the evaluation of the objective function value, and the selection of a robust optimization technique to calculate the optimal value. During the optimization process, the profile is changing continuously until obtaining the optimal horn profile. The main focus of this work was to obtain optimal geometries with the use of few design variables. Two different methods to control the horn profile during the optimization process are used: approximation of the contour of the horn with Hermite polynomials and sinusoidal functions. The numerical results show the efficiency of these methods and it was also found (at least from the engineering point of view) that the solution is not unique to the geometry of the horn to single-frequency. The results for the optimization for more than one frequency are also shown.

11:50

5aNS9. Parametric study of a PULSCO vent silencer. Usama Tohid (Eng., PULSCO, 17945 Sky Park Circle, Ste. G, Irvine, CA 92614, u.tohid@pulsco.com)

We have conducted a parametric study via numerical simulations of a PULSCO vent silencer. The overall objective is to demonstrate the existence of an optimum system performance for a given set of operating conditions by modifying the corresponding geometry of the device. The vent silencer under consideration consists of a perforated diffuser, the silencer body, and a tube module. The tube module consists of a set of tubes through which the working fluid passes. The flow tubes are perforated and surrounded with acoustic packing that is responsible for the attenuation. The mathematical model of the vent silencer is built upon Helmholtz equation for the plane wave solution, and the Delany-Bazley model for the acoustic packing. The geometrical parameters chosen for the parametric study include: the porosity of the diffuser and the flow tubes, the type of packing material used for the tube module, bulk density for the acoustic packing, and the hole diameter of the perforated diffuser and flow tubes. The equations of the mathematical model are discretized over the computational domain and solved with a finite element method. Numerical results in terms of transmission loss, for the system, indicate that diffuser hole size of 1/4" with porosity of 0.1, flow tube hole size of 1/8" with porosity of 0.23, packing density of 16 kg/m^3 for TRS-10 and 100 kg/m^3 for Advantex provided the optimum results for the chosen set of conditions. The numerical results were found to be in agreement with experimental data.

Session 5aPPa

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

Noah H. Silbert, Chair
Communication Sciences & Disorders, University of Cincinnati, 3202 Eden Avenue, 344 French East Building, Cincinnati, OH 45267

All posters will be on display from 8:00 a.m. to 10:00 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:00 a.m. and contributors of even-numbered papers will be at their posters from 9:00 a.m. to 10:00 a.m.

Contributed Papers

5aPPa1. Is age-related hearing loss predominantly metabolic? Robert H. Withnell (Speech and Hearing Sci., Indiana Univ., Bloomington, IN) and Margarete A. Ueberfuhr (Systemic NeuroSci., Ludwig-Maximilians Univ., Großhaderner Str. 2, D-82152 Planegg-Martinsried, Munich, Germany, margarete.ueberfuhr@gmx.de)

Studies in animals have shown that age-related hearing loss is predominantly metabolic in origin. In humans, direct access to the cochlea is not usually possible and so non-invasive methods of assessing cochlear mechanical function are required. This study used a non-invasive assay of cochlear mechanical function, otoacoustic emissions, to examine a metabolic versus hair-cell-loss origin for age-related hearing loss. Three subject groups were examined: adult females with clinically normal hearing, adult females with age-related hearing loss, and adult males with noise-induced hearing loss. Contrasting otoacoustic emission input-output functions were obtained for the three groups, suggesting a causal relationship between age-related hearing loss and strial dysfunction.

5aPPa2. Further modeling of temporal effects in two-tone suppression. Erica L. Hegland and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, ehegland@purdue.edu)

Two-tone suppression, a nearly instantaneous reduction in cochlear gain and a by-product of the active process, has been extensively studied both physiologically and psychoacoustically. Some physiological data suggest that the medial olivocochlear reflex (MOCR), which reduces the gain of the active process in the cochlea, may also reduce suppression. The interaction of these two gain reduction mechanisms is complex and has not been widely studied or understood. Therefore, a model of the auditory periphery that includes the MOCR time course was used to systematically investigate this interaction of gain reduction mechanisms. This model was used to closely examine two-tone suppression at the level of the basilar membrane using suppressors lower in frequency than the probe tone. Results were compared both with and without elicitation of the MOCR. Preliminary results indicate that elicitation of the MOCR reduces two-tone suppression when measured as the total basilar membrane response at the characteristic frequency (CF) of the probe. The purpose of this study was to investigate further by separating the frequency components of the basilar membrane response at CF to determine the excitation produced by the probe and by the suppressor with and without MOCR elicitation. [Research supported by NIH(NIDCD)R01 DC008327 and T32 DC00030.]

5aPPa3. Characterization of cochlear implant-related artifacts during sound-field recording of the auditory steady state response using an amplitude modulated stimulus: A comparison among normal hearing adults, cochlear implant recipients, and implant-in-a-box. Shruti B. Deshpande (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267-0379, balvalsn@mail.uc.edu), Michael P. Scott (Div. of Audiol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), Fawen Zhang, Robert W. Keith (Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH), and Andrew Dimitrijevic (Commun. Sci. Res. Ctr., Cincinnati Children's Hospital, Dept. of Otolaryngol., Univ. of Cincinnati, Cincinnati, OH)

Recent work has investigated the use of electric stimuli to evoke auditory steady state response (ASSR) in cochlear implant (CI) users. While more control can be exerted using electric stimuli, acoustic stimuli present natural listening environment for CI users. However, ASSR using acoustic stimuli in the presence of a CI could lead to artifacts. Five experiments investigated the presence and characteristics of CI-artifacts during sound-field ASSR using amplitude modulated (AM) stimulus (carrier frequency: 2 kHz; modulation frequency: 82.031 Hz). Experiment 1 investigated differences between 10 normal hearing (NH) and 10 CI participants in terms of ASSR amplitude versus intensity and onset phase versus intensity. Experiment 2 explored similar relationships for an implant-in-a-box. Experiment 3 investigated correlations between electrophysiological ASSR thresholds (ASSRe) and behavioral thresholds to the AM stimulus (BTAM) for the NH and CI groups. Mean threshold differences (ASSRe-BTAM) were computed for each group and group differences were studied. Experiment 4 investigated the presence of transducer-related artifacts using masking. Experiment 5 investigated the effect of manipulation of intensity and external components of the CI on the ASSR. Overall, results of this study provide the first comprehensive description of the characteristics of CI-artifacts during sound-field ASSR. Implications for future research to further characterize CI-artifacts, thereby leading to strategies to minimize them are discussed.

5aPPa4. Baseline neurophysiological noise levels in children with auditory processing disorder. Kyoko Nagao (Biomedical Res., Nemours/Alfred I. duPont Hospital for Children, 1701 Rockland Rd., CPASS, Wilmington, DE 19803, knagao@nemours.org), L. Ashleigh Greenwood (Audiol. Services, Pediatrix , Falls Church, VA), Raj C. Sheth, Rebecca G. Gaffney (Biology, Univ. of Delaware, Newark, DE), Matthew R. Cardinale (College of Osteopathic Medicine, New York Inst. of Technol., New York, NY), and Thierry Morlet (Biomedical Res., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE)

The current study examined the baseline neurophysiological responses between children with auditory processing disorder (APD) and the control group. Auditory event related potentials were recorded in 23 children with APD (ages 7–12 years, mean age = 8.9 years) and 25 age-matched control

children in response to a /da/ presented to each ear separately (right and left ear conditions). A no-sound condition was recorded as well. Baseline neurophysiological activity was measured as the root mean square amplitude of the 100 ms pre-stimulus period. Preliminary analysis of data from 19 children with APD and 13 controls indicated that the APD group showed significantly greater pre-stimulus amplitude than the control group in the left ear condition, $F(1, 30) = 4.415$, $p = 0.044$, but we did not find significant group differences in the no-sound and right ear conditions, $F(1, 30) = 2.237$, $p = 0.15$ and $F(1, 30) = 0.088$, $p = 0.77$, respectively. The results suggest that children with APD may need a longer time period to return to a resting state than control children when the left ear is stimulated. Hence, these results may indicate asymmetrical neural activities of the auditory pathways in APD.

5aPPa5. Speech spectral intensity discrimination at frequencies above 6 kHz. Brian B. Monson (Dept. of Pediatric Newborn Medicine, Brigham and Women's Hospital, Harvard Med. School, 75 Francis St., Boston, MA 02115, bmonson@research.bwh.harvard.edu), Andrew J. Lotto, and Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Hearing aids and other communication devices (e.g., mobile phones) have made some recent efforts to extend their bandwidths to represent higher frequencies. The impact of this expansion on speech perception is not well characterized. To assess human sensitivity to speech high-frequency energy (HFE, defined here as energy in the 8- and 16-kHz octave bands), difference limens for HFE level changes in male and female speech and singing were obtained. Listeners showed significantly greater ability to detect level changes in singing vs. speech, but not in female vs. male speech. Mean differences limen scores for speech and singing were about 5 dB in the 8-kHz octave (5.6–11.3 kHz) but 8–10 dB in the 16-kHz octave (11.3–22 kHz). These scores are lower (better) than scores previously reported for isolated vowels and some musical instruments, and similar to scores previously reported for white noise.

5aPPa6. Duration perception of time-varying sounds: The role of the amplitude decay and rise-time modulator. Lorraine Chuen (Psych., Neurosci. & Behaviour, McMaster Univ., Psych. Bldg. (PC), Rm. 102, 1280 Main St. West, Hamilton, ON L8S 4K1, Canada, chuenll@mcmaster.ca) and Michael Schutz (School of the Arts, McMaster Univ., Hamilton, ON, Canada)

It is well known that ramped (rising energy) sounds are perceived as longer in duration than damped (falling energy) sounds that are time-reversed, but otherwise identical versions of one another (Schlauch, Ries & DiGiovanni, 2001; Grassi & Darwin, 2006). This asymmetry has generally been attributed to the under-estimation of damped sound duration, rather than the over-estimation of ramped sound duration. As previous literature most commonly employs exponential amplitude modulators, in the present experiment, we investigate whether altering the nature of this amplitude decay- or rise-time modulator (linear or exponential) would influence this typically observed perceptual asymmetry. Participants performed an adaptive, 2AFC task that assessed the point of subjective equality (PSE) between a standard tone with a constant ramped/damped envelope, and a comparator tone with a "flat," steady state envelope whose duration varied according to a 1-up, 1-down rule. Preliminary results replicated previous findings that ramped sounds are perceived as longer than their time-reversed, damped counterparts. However, for sounds with a linear amplitude modulator, this perceptual asymmetry is partially accounted for by ramped tone over-estimation, contrasting previous findings in the literature conducted with exponential amplitude modulators.

5aPPa7. Relationship between gap detection thresholds and performance on the Advanced Measures of Music Audiation Test. Matthew Hoch (Music, Auburn Univ., Auburn, AL), Judith Blumsack, and Lindsey Soles (CMDS, Auburn Univ., 1199 Haley Ctr., Auburn, AL 36849-5232, blumsjt@auburn.edu)

Considerable neurophysiological, neural imaging, and behavioral research indicates that auditory processing in musicians differs from that of non-musicians (e.g., Musacchia et al., 2007; Ohnishi et al., 2001; Pantev et al., 1998; Parbery-Clark, 2009). Among the auditory skills in musicians that have been studied are gap detection measures of temporal acuity (Mishra & Panda, 2014; Payne, 2012). These studies typically have compared the gap detection thresholds of musicians and non-musicians. The present work relates gap detection performance to musical aptitude rather than to reported musical training history. In addition, in the present study, gap detection was measured under two different stimulus conditions: the within-channel (WC) condition (in which the sound that precedes the gap is spectrally identical to the sound following the gap) and the across-channel (AC condition) (in which the pre- and post-gap sounds are spectrally different. Results indicate a significant correlation between across-channel gap detection thresholds and musical aptitude and no correlation between within-channel performance and musical aptitude. These results have important implications for temporal acuity as it relates to musical aptitude.

5aPPa8. Modeling response times to analyze perceptual interactions in complex non-speech perception. Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., 344 French East Bldg., Cincinnati, OH 45267, noah.silbert@uc.edu) and Joseph W. Houpt (Psych., Wright State Univ., Dayton, OH)

General recognition theory (GRT) provides a powerful framework for modeling interactions between perceptual dimensions in identification-confusion data. The linear ballistic accumulator (LBA) model provides powerful methods for analyzing multi-choice (2+) response time (RT) data as a function of evidence accumulation and response thresholds. We extend (static) GRT to the domain of RTs by fitting LBA models to RTs collected in two auditory GRT experiments. Although the mapping between the constructs of GRT (e.g., perceptual separability, perceptual independence) and the components of the LBA (e.g., drift rates, response thresholds) is complex, the dimensional interactions defined in GRT can be indirectly addressed in the LBA framework by testing for invariance of LBA parameters across appropriate subsets of the data. The present work focuses on correspondences between (invariance of) parameters in LBA and perceptual separability and independence in GRT.

5aPPa9. The effect of experience on environmental sound identification. Rachel E. Bash, Brandon J. Cash, and Jeremy Loebach (Psych., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, bash@stolaf.edu)

The perception of environmental stimuli was compared across normal hearing (NH) listeners exposed to an eight-channel sinewave vocoder and experienced bilateral, unilateral, and bimodal cochlear implant (CI) users. Three groups of NH listeners underwent no training (control), one day of training with environmental stimuli (exposure), or four days of training with a variety of speech and environmental stimuli (experimental). A significant effect of training was observed. The experimental group performed significantly better than exposure or control groups, equal to bilateral CI users, but worse than bimodal users. Participants were divided into low, medium and high-performing groups using a two-step cluster algorithm. High-performing members were only observed for the CI and experimental conditions, and significantly more low-performing members were observed for exposure and control conditions, demonstrating the effectiveness of training. A detailed item-analysis revealed that the most accurately identified sounds were often temporal in nature or contained iconic repeating patterns (e.g., a horse galloping). Easily identified stimuli were common across all groups, with experimental subjects identifying more short or spectrally driven stimuli, and CI users identifying more animal vocalizations. These data demonstrate that explicit training in identifying environmental stimuli improves sound perception, and could be beneficial for new CI users.

5aPPa10. The U.S. National Hearing Test, a 2013–2014 progress report. Charles S. Watson (Res., Commun. Disord. Technol., Inc., CDT, Inc., 3100 John Hinkle Pl, Bloomington, IN 47408, watson@indiana.edu), Gary R. Kidd (Speech and Hearing, Indiana Univ., Bloomington, IN), James D. Miller (Res., Commun. Disord. Technol., Inc., Bloomington, IN), Jill E. Preminger (Surgery, Univ. of Louisville, Louisville, KY), Alex Crowley, and Daniel P. Maki (Res., Commun. Disord. Technol., Inc., Bloomington, IN)

A telephone-administered screening test for sensorineural hearing loss was made publically available in the United States in September 2013. This test is similar to the digits-in-noise test developed by Smits and colleagues in the Netherlands, versions of which are now in use in most European countries and in Australia. The test was initially offered in the United States for a small fee ($8, then $4) but after a year of promotion it became clear that either the fee or the complexity of paying it was inhibiting. During the first month in which the test was subsequently offered free of charge, 31,806 calls were made to the test line, of which 26,507 were completed tests. Analyses of test performance suggest that about 81% of the test takers had at least a mild hearing loss, and 40% had a substantial loss (estimated to be in excess of 45 dB PTA). Follow-up studies are being conducted to determine whether those who failed the test sought a full-scale hearing assessment, and whether those advised to obtain hearing aids did so. [Work funded by Grant No. 5R44DC009719 from the National Institute for Deafness and other Communication Disorders.]

FRIDAY MORNING, 31 OCTOBER 2014 — MARRIOTT 1/2, 10:15 A.M. TO 12:15 P.M.

Session 5aPPb

Psychological and Physiological Acoustics: Perceptual and Physiological Mechanisms, Modeling, and Assessment

Anna C. Diedesch, Chair
Hearing & Speech Sciences, Vanderbilt University, Nashville, TN 37209

Contributed Papers

10:15

5aPPb1. Modest, reliable spectral peaks in preceding sounds influence vowel perception. Christian Stilp and Paul Anderson (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Sensory systems excel at extracting predictable signal properties in order to be optimally sensitive to unpredictable, more informative properties. Studies of auditory perceptual calibration (Kiefte & Kluender, 2008 JASA; Alexander & Kluender, 2010 JASA) showed that when precursor sounds were filtered to emphasize frequencies matching the second formant (F_2) of the subsequent target vowel, vowel perception decreased its reliance on F_2 (predictable cue) and increased reliance on spectral tilt (unpredictable cue). Perceptual calibration occurred when reliable spectral peaks were 20 dB or larger, but findings in profile analysis and spectral contrast detection predict sensitivity to more modest spectral peaks. The present experiments tested identification of vowels varying in F_2 (1000–2200 Hz) and spectral tilt (-12-0 dB/octave), perceptually varying from /u/ to /i/. Listeners first identified vowels in isolation, then following a sentence filtered to add a reliable + 2 to + 15 dB spectral peak centered at F_2 of the target vowel. Changes in perceptual weights (standardized logistic regression coefficients) across sessions were indices of perceptual calibration. Vowel identification weighted F_2 significantly less when reliable peaks were at least + 5 dB, but increases in spectral tilt weights were very modest. Results demonstrate high sensitivity to predictable acoustic properties in the sensory environment.

10:30

5aPPb2. Testing the contribution of spectral cues on pitch strength judgments in normal-hearing listeners. William Shofner (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, wshofner@indiana.edu) and Marisa Marsteller (Speech, Lang. and Hearing Sci., Univ. of Arizona, Tucson, AZ)

When a wideband harmonic tone complex (wHTC) is passed through a noise vocoder, the resulting sounds can have harmonic structures with large peak-to-valley ratios in the spectra, but little or no periodicity strength in the autocorrelation functions. Noise-vocoded wHTCs evoke simultaneous noise percepts and pitch percepts similar to those evoked by iterated rippled noises. We have previously shown that spectral cues do not appear to control behavioral responses of chinchillas to noise-vocoded wHTCs in a stimulus generalization task, but do appear to contribute to pitch strength judgments in normal-hearing listeners for noise-vocoded wHTCs relative to non-vocoded wHTCs. To further test the role of spectral cues, normal-hearing listeners judged the pitch strengths of noise-vocoded wHTCs relative to infinitely-iterated rippled noise (IIRN). Stimuli had harmonic structures with a fixed fundamental frequency of 500 Hz and were presented monaurally at 50 dB SL. Listeners' judgments of pitch strength evoked by vocoded wHTCs were generally consistent with peak-to-valley ratios of the stimuli. In order to reduce spectral cues and resolvability, stimuli were high-passed filtered. Pitch strength judgments of vocoded wHTCs were reduced following high-pass filtering. These findings suggest that spectral cues do contribute to pitch perception in human listeners.

10:45

5aPPb3. The role of onsets and envelope fluctuations in binaural cue use. G. Christopher Stecker and Anna C. Diedesch (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232-8242, g.christopher.stecker@vanderbilt.edu)

Effective localization of real sound sources requires neural mechanisms to accurately extract and represent binaural cues, including interaural time and level differences (ITD and ILD) in the sound arriving at the ears. Many studies have explored the relative effectiveness of these cues, and how that effectiveness varies with the acoustical features of a sound such as spectral frequency and modulation characteristics. In particular, several classic and recent studies have demonstrated relatively greater sensitivity to ITD and ILD present at sound onsets and other positive-going fluctuations of the sound envelope. The results of those studies have clear implications for how spatial cues are extracted from naturally fluctuating sounds such as human speech, and how that process is altered by echoes, reverberation, and competing sources in real auditory scenes. Here, we review the results of several recent studies to summarize and critique the evidence for envelope-triggered

extraction of ITD and ILD across a wide range of spectral frequencies. A number of competing models for cue extraction in fluctuating envelopes are also considered in light of this evidence. [Work supported by NIH R01-DC011548.]

11:00

5aPPb4. Loudness of a multi-tonal sound field, consisting of either one two-component complex sound source or two simultaneous spatially distributed sound sources. Michaël Vannier and Etienne Parizet (Génie Mécanique Conception, INSA-Lyon, Laboratoire Vibrations Acoustique, 13, Pl. Jean Macé, Lyon 69007, France, michael.vannier@insa-lyon.fr)

The aim of the present study is to provide new elements about the perceived loudness of stationary complex sound fields and test the validity of current models under such conditions. The first part consisted in testing the hypothesis according which the directional loudness of a multi-component sound source could be fully explained by the directional loudness of each of its single components. In this way, the directional loudness sensitivities of a two-component complex sound source (third-octave noise bands centered at 1 kHz and 5 kHz) have been measured in the horizontal plane. Despite a previous equalization in loudness of each component to a frontal reference, a small effect of the azimuth angle on loudness still remained, partly disproving the assumption. In a second part, the influence of the spatial distribution of two sound sources on the global loudness was investigated (with the same two narrow-band noises). No effect has been found by Song (2007) for small incidence angles (10 and 30°). The present experiment extends this result for wide incidence angles and so, under highly dichotic listening situations. Finally, all the subjective data have been compared with the predictions from different models of loudness, and the results will be discussed.

11:15

5aPPb5. Computing interaural differences using idealized head models. Tingli Cai, Brad Rakerd, and William Hartmann (Phys. Astronomy, Michigan State Univ., 567 Wilson Rd., East Lansing, MI 48824, hartman2@msu.edu)

The spherical model of the human head, attributable to Lord Rayleigh, accounts for important features of observed interaural time differences (ITD) and interaural level differences (ILD), but it also fails to capture many details. To gain an intuitive understanding of the failures, we computed ITDs and ILDs for a succession of idealized shapes approximating the human head: sphere, ellipsoid, ellipsoid plus cylindrical neck, ellipsoid plus cylindrical neck plus disk torso. Calculations were done as a function of frequency (100–2500 Hz) and for source azimuths from 10 to 90 degrees using finite-element models. The computations were compared to free-field measurements on a KEMAR manikin. The spherical head model approximated many measured interaural features, but the frequency dependence tended to be too flat in both ITD and ILD. The ellipsoidal head produced greater variation with frequency and therefore agreed better with the measurements, reducing the RMS discrepancies in both ITD and ILD by 35%. Adding a neck further increased the frequency variation. Adding the disk torso further improved the agreement, especially below 1000 Hz, decreasing the ITD discrepancy by another 21\%. The evolution of models enabled us to associate details of interaural differences with overall anatomical features. [Work supported by the AFOSR grant 11NL002.]

11:30

5aPPb6. Acoustic reflex attenuation in phon loudness measurements. Julius L. Goldstein (Hearing Emulations LLC, Ariel Premium,Hearing Emulations LLC, 8825 Page Ave., Saint Louis, MO 63114-6105, goldstein.jl@sbcglobal.net)

Equal Loudness-level Contours, ELC(f, L), represent the sound pressure level in dB SPL of tones at frequency f that are perceived by normal-hearing listeners as equally loud as a 1 kHz tone at L dB SPL. Loudness is defined relatively as L phons (Fletcher & Munson, 1933). ELC measurements by Lydolf and Møller (1997) included in the current ISO standard (Suzuki & Takeshima, JASA vol. 116, 2004), show systematic increases in ELC growth rate with loudness above 60 phons and below 1 kHz, which suggests middle-ear attenuation by the acoustic reflex (AR). A steady-state ELC model was assembled including known mechanisms: (1) middle ear transmission modified by a head-related-transfer-function, (2) compressive cochlear amplification (CA) for signaling loudness, (3) a negative feedback model for AR attenuation by CA inputs exceeding AR threshold, and (4) attenuation of pressure-field stimuli by trans-eardrum static pressure. Model parameters were calculated from ELC data using minimum-square-error estimation. AR attenuation below 1 kHz depends on AR attenuation at the 1 kHz loudness reference frequency, but predicted ELCs are relatively insensitive to it. An earlier psychophysical study of AR attenuation, including 1 kHz, is consistent with subject-dependent model predictions (Rabinowitz & Goldstein, JASA vol. 54, 1973; Rabinowitz, 1977). [NIH-Funded.]

11:45

5aPPb7. Effects of tinnitus and hearing loss on functional brain networks involved in auditory and visual short-term memory. Fatima T. Husain, Kwaku Akrofi (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, husainf@illinois.edu), and Jake Carpenter-Thompson (Neurosci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Brain imaging data were acquired from three subject groups—persons with hearing loss and tinnitus (TIN), individuals with similar hearing loss without tinnitus (HL) and those with normal hearing without tinnitus (NH)—to test the hypothesis that TIN and control subjects use different functional brain networks for short-term memory. Previous studies have provided evidence of a link between hearing disorders such as tinnitus and the reorganization of auditory and extra-auditory functional networks. Greater knowledge of this reorganization could lead to the development of more effective therapies. Data analysis was conducted on fMRI data obtained while subjects performed short-term memory tasks with low or high attentional loads, using both auditory and visual stimuli in separate scanning sessions. Auditory stimuli were pure tones with frequencies between 500 and 1000 Hz. Visual stimuli were Korean fonts, unfamiliar to the subjects. We found similar behavioral response across the three groups for both modalities and tasks. However, the groups differed in their brain response, with these differences being more marked for the auditory tasks and not for the tasks involving visual stimuli.

12:00

5aPPb8. Preliminary results of a two-interval forced-choice method for assessing infant hearing sensitivity. Lynne Werner (Speech & Hearing Sci., Univ. of Washington, 1417 North East 42nd St., Seattle, WA 98105-6246, lawerner@u.washington.edu)

Current methods for assessing infants' hearing are yes-no, single interval procedures. Although bias-free statistics can be used to describe the results of such procedures, with the limited number of trials typically available from an individual infant, use of these statistics can be problematic. A two-interval forced choice method based on infants' anticipatory eye movements toward an interesting visual event is currently under development. Preliminary results indicate that a high proportion of both 3- and 7-month-old infants achieve over 80% correct in the detection of a 70 dB SPL 1000 Hz tone presented through an insert earphone. Infants continue to perform better than expected by chance at levels as low as 25 dB SPL. Thus, a test method based on infant eye movements holds potential as an efficient behavioral method for assessing infant hearing.

Session 5aSC

Speech Communication: Speech Perception and Production in Challenging Conditions (Poster Session)

Alexander L. Francis, Chair
Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

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

5aSC1. A new dual-task paradigm to assess cognitive resources utilized during speech recognition. Andrea R. Plotkowski and Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47906, mitche99@purdue.edu)

Listening to ongoing conversations in challenging situations requires explicit use of cognitive resources to decode and process spoken messages. Traditional speech recognition tests are insensitive measures of this cognitive effort, which may differ greatly between individuals or listening conditions. Furthermore, most dual-task paradigms that have been devised for this purpose generally rely on secondary tasks like reaction time and recall that do not reflect real-world listening demands. A new task was designed to capture changes in both speech recognition and verbal processing across different conditions. Listeners heard two sequential sentences spoken by opposite gender talkers in speech-shaped noise. The primary task was a traditional speech recognition test, in which listeners immediately repeated aloud the second sentence in the pair. The secondary task was designed to engage explicit cognitive processes by requiring listeners to write down the first sentence after holding it in memory while listening to and repeating back the second sentence. Test sentences consisted of lists from the PRESTO test (Gilbert *et al.* 2013, J. Am. Acad. Audiol. vol. 24, pp. 26–36) that were carefully modified to help ensure list-equivalency. Psychometric results from the revised PRESTO sentence lists and from the new dual-sentence task will be reported.

5aSC2. Vocal effort, coordination, and balance. Robert A. Fuhrman (Linguist, U Br. Columbia, 2613 West Mall, Vancouver, BC, Canada, robert.a.fuhrman@gmail.com), Adriano Barbosa (Electron. Eng., Federal Univ. of Minas Gerais, Belo Horizonte, Brazil), and Eric Vatikiotis-Bateson (Linguist, U Br. Columbia, Vancouver, BC, Canada)

Manipulating speaking and discourse requirements allows us to asses the time-varying correspondences between various subsystems within a talker at different levels of vocal effort. These subsystems include fundamental frequency (F0) and acoustic amplitude, rigid body (6D) motion of the head, motion (2D) of the body, and postural forces and torques measured at the feet. Analysis of six speakers has confirmed our hypothesis that as vocal effort increases coordination among sub-systems simplifies, as shown by greater correspondence (e.g., the instantaneous correlation) between the various time-series measures. However, at the two highest levels of vocal effort, elicited by having talkers shout to and yell at someone located appropriately far away, elements of the postural force, notably one or more torque components, often show a reduction in correspondence with the other measures. We interpret this result as evidence that talkers become more rigidly coordinated at the highest levels of vocal effort, which can interfere with their balance. Furthermore, the discourse type—shouting at someone to carry on a conversation vs. yelling at someone not expected to reply—can be associated with differing amounts of imbalance.

5aSC3. The gradient effect of transitional magnitude: A source of the vowel context effect. SANG-IM LEE-KIM (Linguist, New York Univ., 45-35 44th St. 1i, Sunnyside, NY 11104, sangim119@gmail.com)

Previous studies have shown that vocalic transitions play an important role in the identification of the consonantal places (e.g., Whalen 1981/1991, Nowak 2006, Babel & McGuire 2013). While it has been intermittantly reported that the contribution of transitions may depend on vowel contexts, the common methodology, i.e., C-V cross-splicing, is too coarse to precisely identify the nature of this effect. In the present study, vocalic transitions are systematically manipulated and used as a gradient variable by incrementally removing the transitional period of the three vowels /u a e/ following alveolopalatal sibilant /ɕ/ in Polish. In an identification task, native Polish speakers were given a choice between /ɕ/ and /ʂ/ for stimuli with varying levels of palatal transitions. The results showed that participants' perception is gradient: greater transitions overall elicit more palatal responses in all vowel contexts. More importantly, it has been shown that the apparent vowel effect can be largely reduced to the relative magnitude of transitions that are specific to each vowel. The low and back vowels elicit greater palatal transitions providing more robust transitional cues in perception, while the high and front vowels elicit smaller or nearly zero palatal transitions providing less robust cues to the sibilants' place.

5aSC4. Adaptive compensation for reliable spectral characteristics of a listening context in vowel perception. Paul Anderson and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 2301 S 3rd St., Louisville, KY, paul.anderson@louisville.edu)

When precursor sounds are filtered to emphasize frequencies matching F_2 of a subsequent target vowel, vowel perception decreases reliance on F_2 (predictable cue) and increases reliance on spectral tilt (unpredictable cue) and vice versa. Previously, initial cue weights and weight changes (i.e., perceptual calibration to reliable signal properties) were larger for F_2 than tilt, obscuring whether the magnitude of calibration reflects cue predictability or F_2's status as a primary cue to vowel identity. Here, vowels varied from /u/ to /i/ in tilt (-12-0 dB/octave) and the full range of F_2 values (1000–2200 Hz) or a reduced range (1300–1900 Hz) designed to decrease F_2 cue weights, making tilt the primary cue for vowel identification. Vowels were presented in isolation, then following sentences filtered to match the target vowel's F_2 or tilt. In isolation, cue weights for F_2 were higher when identifying full-F_2-range vowels and higher for tilt when identifying reduced-F_2-range vowels. Weight changes (calibration) were comparable when the primary cue was predictable; this was also true for predictable secondary cues (tilt for full-F_2-range vowels, F_2 for reduced-F_2-range vowels). Perceptual calibration to reliable signal properties is an adaptive process reflecting cue predictability, not solely *a priori* cue use (e.g., F_2 over tilt).

5aSC5. An approach to the analysis of relations between syllable and sentence perception in quiet and noise in the Speech Perception Assessment and Training System: Preliminary results for ten hearing-aid users. James D. Miller (Res., Commun. Disord. Technol., Inc., 3100 John Hinkle Pl, Ste 107, Bloomington, IN 47408, jamdmill@indiana.edu)

Logistic functions relating abilities to indentify syllable onsets, nuclei, and codas in quiet and noise as a function of snr are measured. Syllable perception is the product of these individual abilities. It is found that syllable perception in noise is highly correlated with syllable perception quiet. The relation of sentence perception in the SPATS sentence task with SPATS syllable constituent perception is examined. As shown years ago at the Bell Labs, only modest levels of syllable identification are needed to support nearly perfect levels of sentence perception. Here, it is found that sentence perception in quiet and noise is correlated with syllable perception in quiet, the use of inherent context provided by syllable perception (Boothroyd and Nittrauer (1988)), and with the use of situational context, independent of syllable perception. Finally, the effects of speech perception training on these relations are examined for each of the ten hearing-aid users studied. [Work supported by NIH/NICD Grant R21/R33DC011174 "Multi-site Study of the Efficacy of Speech Perception Training for Hearing-Aid Users," C. S. Watson, PI. Data supplied by cooperating sites: Medical University of South Carolina, J. Dubno, Site PI; University of Memphis, D. Wark, Site PI; and University of Maryland, S. Gordon-Salant, Site PI.]

5aSC6. Neural-scaled entropy predicts the effects of nonlinear frequency compression on speech perception. Varsha Hariram and Joshua Alexander (Speech Lang. and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, vhariram@purdue.edu)

Signal processing schemes used in hearing aids, such as nonlinear frequency compression (NFC) recode speech information by moving high-frequency information to lower frequency regions. Perceptual studies have shown that depending on the dominant speech sound, compression occurs and the amount of compression can have a significant effect on perception. Very little is understood about how frequency-lowered information is encoded by the auditory periphery. We have developed a measure that is sensitive to information in the altered speech signal in an attempt to predict optimal hearing aid settings for individual hearing losses. The Neural-Scaled Entropy (NSE) model examines the effects of frequency-lowered speech at the level of the inner hair cell synapse of an auditory nerve model [Zilany *et al.* 2013, Assoc. Res. Otolaryngol.]. NSE quantifies the information available in speech by the degree to which the pattern of neural firing across frequency changes relative to its past history (entropy). Nonsense syllables with different NFC parameters were processed in noise. Results are compared with perceptual data across the NFC parameters as well as across different vowel-defining parameters, consonant features, and talker gender. NSE successfully captured the overall effects of varying NFC parameters across the different sound classes.

5aSC7. Tempo-based segregation of spoken sentences. Gary R. Kidd and Larry E. Humes (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, kidd@indiana.edu)

The ability to make use of differences in speech rhythms to selectively attend to a single spoken message in a multi-talker background was examined in a series of studies. Sentences from the coordinate response measure corpus provided a set of stimuli with a common rhythmic framework spoken by several talkers at similar speaking rates. Subjects were asked to identify two key words spoken in a "target" sentence identified by a word (call sign) near the beginning of the sentence. The target talker was always in the same male voice and either two or six background talkers were presented in different voices (half male and half female). The rate of the background talkers was manipulated to create natural sounding speech that preserved the original pitch and speech rhythms at faster and slower speaking rates. Unaltered target sentences were presented in the presence of faster, unaltered, or slower competing sentences. Performance was poorest with matching target and background tempos, with substantial increases in performance as the tempo differences increased. Modification of the target-sentence rate confirmed that the effect is due to the relative timing of target and background speech, rather than the properties of rate-modified background speech. [Work supported by NIH-NIA.]

5aSC8. Perceptual versus cognitive speed in a time-compressed speech task. Michelle R. Molis, Frederick J. Gallun, and Nirmal Srinivasan (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov)

Time-compression retains the information-bearing spectral change present uncompressed speech, although at a rate that may outstrip cognitive processing speed. To compare the relative importance of perceptual and cognitive processing speed, we compared the understanding of (1) time-compressed stimuli expanded in time via gaps with (2) uncompressed stimuli where spectral change information was removed. We hypothesized that, despite the initial compression, the compressed and expanded stimuli would be more intelligible as it would retain relatively more information-bearing spectral change. Participants were somewhat older listeners (mid-1950s to mid-1960s) with normal hearing or mild hearing loss. Stimuli were spoken seven-digit strings time-compressed via pitch synchronous overlap and add (PSOLA) at three uniform compression ratios (2:1, 3:1, and 5:1). In gap insertion conditions, the total duration of the compressed stimuli was restored via introduction of periodic gaps. This produced signal-to-gap ratios of 1:1, 1:2, and 1:4. For comparison, segments of unaccelerated strings, equal to the duration of the inserted gaps, were zeroed out resulting in the same signal-to-gap ratios. Listeners identified the final four digits of the strings presented in quiet and in a steady-state, speech-shaped background noise (SNR +5). Our hypothesis was supported for the fastest compression rates. [Work supported by VA RR&D.]

5aSC9. Information-bearing acoustic changes are important for understanding vocoded speech in a simulation of cochlear implant processing strategies. Christian Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Information-bearing acoustic changes (IBACs) in the speech signal are important for understanding speech. This was demonstrated with cochlea-scaled entropy for cochlear implants (CSE_{CI}), which measures perceptually significant intervals of noise-vocoded speech (Stilp *et al.*, 2013 JASA; Stilp, 2014 JASA; Stilp & Goupell, 2014 ASA). However, vocoding does not necessarily mimic CI processing. Some CI processing strategies present acoustic information in all channels at all times (e.g., CIS) while others present only the *n*-highest-amplitude channels out of *m* at any time (e.g., ACE). Here, IBACs were explored in a simulation of ACE processing. Sentences were divided into 22 channels spanning 188–7938 Hz and noise-vocoded. In each 1-ms interval (simulating 1000 pulses/second stimulation rate), only the eight highest-amplitude channels were retained. CSE_{CI} was calculated between 1-ms or 16-ms sentence segments, then summed into 80-ms intervals. High-CSE_{CI} or low-CSE_{CI} intervals were replaced by speech-shaped noise. Consistent with previous studies, replacing high-CSE_{CI} intervals impaired sentence intelligibility more than replacing an equal number of low-CSE_{CI} intervals. Importantly, performance was comparable when 1- or 16-ms IBACs were replaced by noise. Results reveal the perceptual importance of IBACs on rapid timescales after simulated ACE processing, indicating this information is likely available to CI users for understanding speech.

5aSC10. Talker intelligibility across clear and sinewave vocoded speech. Jeremy Loebach, Gina Scharenbroch, and Katelyn Berg (Psych., St. Olaf College, 1520 St. Olaf Ave., Northfield, MN 55057, loebach@stolaf.edu)

Talker intelligibility was compared across clear and sinewave vocoded speech. Ten talkers (5 female) from the Midwest and Western dialect regions recorded samples of 210 meaningful IEEE sentences, 206 semantically anomalous sentences, and 300 MRT words. Ninety-three normal hearing participants provided open set transcriptions of the materials presented in the clear over headphones. Forty-one different normal hearing participants provided open set transcriptions of the materials processed with an eight-channel sinewave vocoder. Transcription accuracy was highest for clear speech compared to vocoded speech, and for meaningful sentences, followed by anomalous sentences and words for both conditions. Weak talker effects were observed for the meaningful sentences in the clear (ranging from 97.7% to 98.2%), but were more pronounced for vocoded versions

(68.5% to 85.5%). Weak talker effects were observed for semantically anomalous sentences in the clear (89.4%-93.3%), but more variability was observed across talkers in the vocoded condition (54.4%–73.7%). Finally, stronger talker effects were observed for clear and vocoded MRT words (83.8%–95.6%, 46.3%–59.0%, respectively). Talker rankings differed across stimulus conditions, as well as across processing conditions, but significant positive correlations between conditions were observed for meaningful and anomalous sentences, but not MRT words. Acoustic and dialect influences on intelligibility will be discussed.

5aSC11. Vowels of four-year-old children with cerebral palsy in Mandarin-learning environment. Li-mei Chen (Foreign Lang. and Lit., National Cheng Kung Univ., Tainan, Taiwan), Yu Ching Lin (Physical Medicine and Rehabilitation, National Cheng Kung Univ., Tainan, Taiwan), Wei Chen Hsu, and Meng-Hsin Yeh (Foreign Lang. and Lit., National Cheng Kung Univ., 1 University Rd, Tainan 701, Taiwan, myonaa@gmail.com)

Characteristics of vowel productions of children with cerebral palsy (CP) were investigated with data from two 4-year-old children with CP and two typically-developing (TD) children in Mandarin-learning environment. Clear vowel productions from picture naming and natural conversation in three 50-minute audio recordings of each child were transcribed and analyzed. Seven parameters were examined: vowel duration of /a/, F2 slope in transition of CV sequence, cumulative change of F2 for vowel /a/, degree of nasalization in oral vowel (A1-P1), percent of jitter, percent of shimmer, and the signal to noise ratio (SNR). Major findings are: (1) The CP group showed shorter vowel duration of /a/; (2) TD group has larger F2 slope in CV transition; (3) No obvious differences were found between TD and CP groups in cumulative change of F2 for vowel /a/, degree of nasalization (A1-P1), and voice perturbation (percent of jitter, percent of shimmer, and SNR). Further study with more participants and with careful data selection can verify findings of this study in search for valid parameters to characterize vowel production of children with CP.

5aSC12. Effects of depression on speech. Saurabh Sahu and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland College Park, 8125 48 Ave., Apt 101, College Park, MD 20740, ssahu89@umd.edu)

In this paper, we are investigating the effects of depression on speech. The motivation comes from the fact that neuro-physiological changes associated with depression affect motor coordination and can disrupt the articulatory precision in speech. We use the database collected by Mundt *et al*. (J. Neurolinguist. vol. 20, no. 1, pp. 50–64, Jan. 2007) in which 35 subjects were treated over a 6 week period and study how the changes in mental state are manifest in certain acoustic properties that correlate with the Hamilton Depression Rating Scale (HAM-D), which is a clinical assessment score. We look at features such as the modulation frequencies, aperiodic energy during voiced speech, vocal fold jitter and shimmer, and other cues that are related to articulatory precision. These measures will be discussed in detail.

5aSC13. Pitch production of a Mandarin-learning infant with cerebral palsy. Meng-Hsin Yeh, Li-mei Chen (Foreign Lang. and Lit., National Cheng Kung Univ., 1 University Rd., Tainan 701, Taiwan, myonaa@gmail.com), Chyi-Her Lin, Yuh-Jyh Lin, and Yung-Chieh Lin (Pediatrics, National Cheng Kung Univ., Tainan, Taiwan)

In this study, pitch production were investigated in two Mandarin-learning infants at 6 months of age, an infant with cerebral palsy (CP) and a typically developing (TD) infant. Words with distinct tones in Mandarin differ in meaning. In order to produce a correct tone, having good control of the respiratory and the laryngeal mechanisms are necessary. Thus, producing a correct tone and reaching intelligibility for children with CP is considered to be relatively difficult. In previous studies, Kent and Murray (1982) pointed out that falling contours predominated in infant vocalizations at 3, 6, and 9 months. A study by Chen et al (2013) with 4-year-old children indicated that the mean pitch duration of CP children is 1.3–1.8 times longer than TD counterparts. In adults, Jeng, Weismer, and Kent (2006) found that the pitch slopes of Mandarin in CP adults are smaller than in healthy adults. Three measures were employed in this current study and the major findings are: (1) Both TD and CP infants produced more falling than rising pitch; (2) The mean duration of pitch in CP is 2.3 times longer than that of TD; (3) The pitch slope in CP is smaller than that of TD.

5aSC14. Linear and non-linear acoustic voice analysis of Persian speaking Parkinson's disease patients. Fatemeh Majdinasab (Speech Therapy, Tehran Univ. of Medical Sci., Tehran, Iran), Maryam Mollashahi, Mansour Vali (Medical Eng., k.N. toosi Univ. of Technol., Tehran, Iran), and Hedieh Hashemi (Dept. of Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH, hashemihedieh@yahoo.com)

Purpose: Many studies have analyzed acoustic voice characteristics (AVC) of Parkinson's disease patients (PDP) by linear or non-linear methods. The aim of this study is to compare the linear and non-linear approaches in acoustic voice analysis of Persian speaking PDPs. Method: This cross sectional, non-experimental study was done on 27 (15 males, 12 females) PDP and 21 healthy age-sex matched subjects (11 males, 10 females). Patients were chosen from attendants of movement disorders clinic using convenience sampling. All of patients evaluated in "on" medication period. AVC consisting average fundamental frequency (f0), standard deviation of f0, mean percentage of jitter, shimmer, and HNR in prolongation of all Persian vowels /a, e, i, o, u /. PRAAT 5.1.17 software (as a linear tool) and MATLAB (as a non-linear method) used to evaluate AVC. Result: There was not any significant difference between PDPs and normal subjects except for jitter /æ /(0.041) and / e/ (0.021). According to non-linear characteristics of Wavelet entropy coefficient, and by mother wavelet with coif1 (in MATLAB), all of AVC of patients differentiated from normal. Conclusion: It seems that non-linear analysis is more detailed method to discriminate dysarthric voice from normal voice. Keywords: Acoustic voice analysis, Parkinson's disease, linear, nonlinear.

5aSC15. Vowel development in children with Down and Williams syndromes. Ewa Jacewicz, Robert A. Fox (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu), Vesna Stojanovik, and Jane Setter (Dept. of Clinical Lang. Sci., Univ. of Reading, Reading, United Kingdom)

Down syndrome (DS) and Williams syndrome (WS) are genetic disorders resulting from different types of genetic errors. While both disorders lead to phonological and speech motor deficits, particularly little is known about vowel production in DS and WS. Recent work suggests that impaired vowel articulation in DS likely contributes to the poor intelligibility of DS speech. Developmental delays in temporal vowel structure and pitch control have been found in children with WS when compared to their chronological matches. Here, we analyze spontaneous speech samples produced by British children with DS and WI and compare them with typically developing children from the same geographic area in Southern England. We focus on the acquisition of fine-grained phonetic details, asking if children with DS and WS are able to synchronize the phonetic and indexical domains while coping with articulatory challenges related to their respective syndromes. Phonetic details pertaining to the spectral (vowel-inherent spectral change) and indexical (regional dialect) vowel features are examined and vowel spaces are derived from formant values sampled at multiple temporal locations. Variations in density patterns across the vowel space are also considered to define the nature of the acoustic overlap in vowels related to each syndrome.

5aSC16. Prosodic characteristics in young children with autism spectrum disorder. Laura Dilley, Sara Cook, Ida Stockman, and Brooke Ingersoll (Michigan State Univ., Dept. of Communicative Sci., East Lansing, MI 48824, ldilley@msu.edu)

The prosody of high-functioning adults and adolescents with autism spectrum disorder (ASD) has been reported to differ from that of typically developing individuals. The present study investigated whether young children under eight years old with ASD differ in prosodic characteristics compared with neurotypical children matched on expressive language ability. Seven children with ASD (38–93 months) and seven neurotypical children (20–30 months) were recorded during naturalistic interactions with a parent. Naïve listeners (n = 18) were recruited to rate utterances for: (i) age, (ii)

percentage of intelligible words, (iii) pitch, (iv) speech rate, (v) degree of animation, and (vi) certainty of diagnosis. An acoustic analysis was also conducted of speech rate and fundamental frequency (F0). Results of the rating task showed no statistically significant difference on any measure except estimated age. However, children in the ASD group had a significantly lower mean, maximum, and minimum F0 than children in the control group; there was no significant difference between groups for speech rate. These findings may indicate that speech characteristics alone are unlikely to be a sufficient early sign of an ASD diagnosis.

5aSC17. Speech production changes and intelligibility with a real-time cochlear implant simulator. Lily Talesnick (Neurosci., Trinity College, 300 Summit St., Hartford, CT 06106, lily.talesnick@trincoll.edu) and Elizabeth D. Casserly (Psych., Trinity College, Hartford, CT)

Subjects hearing their speech through a real-time cochlear implant (CI) simulator alter their production in multiple ways, e.g., reducing speaking rate and constricting F1/F2 vowel space. The motivations behind these alterations, however, are currently unknown. Two possibilities are that the changes in speech are due to the influence of a direct feedback loop in which the subject is adjusting speech production to minimize acoustic "error," or that the changes could reflect the indirect influence of a high cognitive load (stemming from the challenge of hearing through the real-time CI simulator). We explored these two possibilities by conducting a playback experiment in which 35 naïve listeners assessed the intelligibility of speech produced under conditions of normal versus vocoded feedback. Intelligibility of vocoded isolated word stimuli in each condition was tested in both a two-alternative forced choice task ("Which recording is easier to understand?") and an open-set word recognition task. Listeners found normal-feedback speech significantly more intelligible in both tasks (p's < 0.0125), suggesting that speakers were not adjusting for direct error correction, but rather due to the influence of an intervening factor, e.g., high cognitive load. Confusion matrix analyses further illuminate the perceptual consequences of the effects of CI-simulated speech feedback.

5aSC18. Hearing and hearing-impaired children's acoustic–phonetic adaptations to an interlocutor with a hearing impairment. Sonia Granlund, Valerie Hazan (Speech, Hearing & Phonetic Sci., Univ. College London (UCL), Rm. 326, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, s.granlund@ucl.ac.uk), and Merle Mahon (Developmental Sci., Univ. College London (UCL), London, United Kingdom)

In England, the majority of children with a hearing impairment attend mainstream schools. However, little is known about the communication strategies used by children when interacting with a peer with hearing loss. This study examined how children with normal-hearing (NH) and those with a hearing impairment (HI) adapt to the needs of a HI interlocutor, focusing on the acoustic–phonetic properties of their speech. Eighteen NH and 18 HI children between the ages of 9 and 15 years performed two problem-solving communicative tasks in pairs: one session was completed with a friend with normal hearing (NH-directed speech) and one session was done with a friend with a hearing impairment (HI-directed speech). As expected, task difficulty increased in interactions involving a HI interlocutor. HI speakers had a slower speech rate, higher speech intensity, and greater F0 range than NH speakers. However, both HI and NH participants decreased their speech rate, and increased their F0 range, mean F0 and the intensity of their speech in HI-directed speech compared to NH-directed speech. This suggests that both NH and HI children are able to adapt to the needs of their interlocutor, even though speech production is more effortful for HI children than their NH peers.

5aSC19. Objective speech intelligibility prediction in sensorineural hearing loss using acoustic simulations and perceptual speech quality measures. Emma Chiaramello, Stefano Moriconi, and Gabriella Tognola (Inst. of Electronics, Computers and TeleCommun. Eng., CNR Italian National Res. Council, Piazza Leonardo Da Vinci 32, Milan 20133, Italy, gabriella.tognola@ieiit.cnr.it)

A novel approach to objectively predict speech intelligibility in sensorineural hearing loss using acoustic simulations of impaired perception and objective measures of perceptual speech quality (PESQ) is proposed and validated. Acoustic simulations of impaired perception with different types and degrees of hearing loss were obtained degrading of the original speech waveforms by spectral smearing, expansive nonlinearity, and level scaling. The CUNY NST syllables were used as test material. PESQ was used to measure perceptual quality of the acoustic simulations thus obtained. Finally, PESQ scores were transformed into predicted intelligibility scores using a logistic function. Validation of the proposed objective method was performed by comparing predicted intelligibility scores with subjective measures of intelligibility of the degraded speech in a group of ten subjects. Predictive intelligibility scores showed good correlation (R² = 0.7) with subjective intelligibility scores and a low error in the prediction (RMSE = 0.14). The proposed approach could be a valuable aid in real clinical applications where it is needed to measure speech intelligibility and might be of some help in avoiding time-consuming experimental measurements. In particular, this method might be valuable in the characterization of the sensitivity of new speech tests for screening and diagnosing of hearing loss, or in the assessment of the performance of novel algorithms of speech enhancement for a target hearing impairment.

5aSC20. Identification of dialect cues by dyslexic and non-dyslexic listeners. Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu), Gayle Long, and Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Spoken language encodes two different forms of information: linguistic (related to the message) and indexical (e.g., speaker's age, gender, and regional dialect). However, some speech-language impairments (such as dyslexia) can reduce a listener's ability to process both linguistic and indexical speech cues. For example, Perrachione *et al.* (Science, 333, 2011) demonstrated that individuals with dyslexia were less able to identify new voices than were control listeners. This study examines the ability of listeners with and without dyslexia to identify speaker dialect. Eighty listeners—40 adults and 40 children (20 in each group were dyslexic, 20 were not; 40 were males and 40 were females)—listened to a set of 80 sentences produced by English speakers from Western North Carolina or central Ohio and were asked to identify which region the speaker came from. Results demonstrated that adult listeners were significantly better at dialect identification and that listeners with dyslexia were significantly poorer at dialect identification. More notably, there was a significant age by listener group interaction—the improvement in dialect identification between adults and children was significantly smaller in listeners with dyslexia. This indicates that an initial limitation in language learning can inhibit long-term development of speaker-specific phonetic representations.

5aSC21. Individual differences in the lexical processing of phonetically reduced speech. Rory Turnbull (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, turnbull@ling.osu.edu)

There is widespread evidence that phonetically reduced speech is processed slower and more effortfully than unreduced speech. However, individual differences in degree and strategies of reduction, and their effects on lexical access, are largely unexplored. This study explored the role of autistic traits in the production and perception of reduced pronunciation variants. Stimuli were recordings of words produced in either high reduction (HR) or low reduction (LR) contexts, extracted from sentences produced by talkers ranging in autism-spectrum quotient (AQ) scores. The reductions in these stimuli were generally small temporal differences, rather than segmental-level alterations such as /t/-flapping. Listeners completed a lexical decision task with these stimuli and the autism-spectrum quotient (AQ) questionnaire. Confirming previous research, the results demonstrate that response times (RTs) to reduced words were slower than to unreduced words. No other effects on RT were observed. In terms of response accuracy, LR words were responded to more accurately than HR words, but this pattern was only observed for temporally reduced words. This LR word accuracy benefit was larger for listeners with more autistic personality traits. These results suggest that individuals differ in the extent to which unreduced speech provides a perceptual benefit.

5aSC22. Change of static characteristics of Japanese word utterances with aging. Mitsunori Mizumachi and Kazuto Ogata (Dept. of Elec. Eng. and Electronics, Kyushu Inst. of Technol., 1-1 Sensui-cho, Tobata-ku, Kitakyushu, 805-8440, Japan, mizumach@ecs.kyutech.ac.jp)

Acoustical characteristics of elderly speech have been investigated in the various viewpoints. Elderly speech can be subjectively characterized by roughness, breathiness, asthenic, and hoarseness. Those characteristics have been individually explained in both medical science and speech science. In particular, the hoarseness, which is caused by a physiological problem with an aged vocal cord, is the most well-known static properties of elderly speech. Change of the hoarseness is quantitatively investigated with aging. Japanese phonetically-balanced 543 word utterances were collected with the cooperation of 153 speakers, whose ages ranged from 20 to 89 years old. Acoustical characteristics of the word utterances were examined in the viewpoints of age and auditory impression. In the static acoustical analysis of Japanese vowels /a/, /e/, /i/, /o/, and /u/, it is confirmed that energy in the high frequency region rises with aging. There is a remarkable energy lift over 4 kHz, and the amount of the energy lift is proportion to the degree of subjective hoarseness.

5aSC23. Effect of formant characteristics on older listeners' dynamic pitch perception. Jing Shen (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, jing.shen@northwestern.edu), Richard Wright (Linguist, Univ. of Washington, Seattle, Washington), and Pamela Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Previous research suggested a large inter-subject variability in dynamic pitch perception among older individuals (Souza *et al.*, 2011). Although data from younger listeners with normal hearing indicate temporal and spectral variations in complex formant characteristics may influence dynamic pitch perception (Green *et al.*, 2002), the present study examines this interaction in an aging population. The stimulus set includes two monophthongs that have static formant patterns and two diphthongs that have dynamic formant patterns. The fundamental frequency at the midpoint in time of each vowel is kept consistent, while the ratio of start-to-end frequency varies in equal logarithmic steps. Older adults with near-normal hearing are tested using an identification task, in which they are required to identify the pitch glide as either "rise" or "fall." An experimental task of AX discrimination is also included to verify the identification data. Results to date show inter-subject variability in dynamic pitch perception among listeners with good static pitch perception. Better pitch glide perception with monophthong than diphthong is observed in those individuals who perform poorly in general. The findings suggest a connection between individual abilities to perceive dynamic pitch and to extract the cues from fundamental and formant frequencies. [Work supported by NIH.]

5aSC24. Sentence recognition in older adults. Kathleen F. Faulkner (Dept. of Psychol. and Brain Sci., Indiana Univ., 1101 E 10th St., Bloomington, IN 47401, katieff@indiana.edu), Gary R. Kidd, Larry E. Humes (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), and David B. Pisoni (Dept. of Psychol. and Brain Sci., Indiana Univ., Bloomington, IN)

Many older adults report difficulty when listening to speech in background noise. These difficulties may arise from some combination of factors, including age-related hearing loss, auditory sensory processing difficulties, and/or general cognitive decline. To perform well in everyday noisy environments, listeners must quickly adapt, switch attention, and adjust to multiple sources of variability in both the signal and listening environments. Sentence recognition tests in noise have been useful for assessing speech understanding abilities because they require a combination of basic sensory/perceptual abilities as well as cognitive resources and processing operations. This study was designed to explore several factors underlying individual differences in aided speech understanding in older adults. We examined the relations between measures of speech perception, cognition, and self-reported listening difficulties in a group of aging adults (N = 40, age range 60–86) and a group of young normal hearing listeners (N = 28, age range 18–30). All participants completed a comprehensive battery of tests, including cognitive, psychophysical, speech understanding, as well as the SSQ self-report scale. While controlling for audibility, speech understanding declined with age and was strongly correlated with psychophysical measures, cognition, and self-reported speech understanding difficulties. [Work supported by NIH: NIDCD grant T32-DC00012 and NIA grant R01-AG008293 to Indiana University.]

5aSC25. Individual differences in speech perception in noise: A neurocognitive genetic study. Zilong Xie (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, 2504A Whitis Ave. (A1100), Austin, TX 78712, xzilong@gmail.com), W. Todd Maddox (Dept. of Psych., The Univ. of Texas at Austin, Austin, TX), Valerie S. Knopik (Div. of Behavioral Genetics, Rhode Island Hospital, Brown Univ. Med. School, Providence, RI), John E. McGeary (Providence Veterans Affairs Medical Ctr. , Providence, RI), and Bharath Chandrasekaran (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, Austin, TX)

Previous work has demonstrated that individual listeners vary substantially in their ability to recognize speech in noisy environments. However, little is known about the underlying sources of individual differences in speech perception in noise. Noise varies in the levels of energetic masking (EM) and informational masking (IM) imposed on target speech. Relative to EM, release from IM places greater demand on selective attention. A polymorphism in exon III of the DRD4 gene has been shown to influence selective attention. Here we investigated whether this polymorphism contributes to individual variation in speech recognition ability. We assessed sentence recognition performance across a range of maskers (1-, 2-, and 8-talker babble, and speech-spectrum noise) among 104 young, normal-hearing adults. We also measured their working memory capacity with Operation Span Task, which relies on selective attention to update and maintain items in memory while performing a secondary task. Results showed that the long variant of the DRD4 gene significantly associated with better recognition performance in 1-talker babble conditions only, and that this relation was mediated by enhanced working memory capacity. These findings suggest that the DRD4 polymorphism can explain some of the individual differences in speech recognition ability, but is specific to IM conditions.

5aSC26. Potential sports concussion identification using acoustic-phonetic analysis of vowel productions. Terry G. Horner (Indiana Univ. Methodist Sports Medicine, 201 Pennsylvania Parkway, Ste. 100, Indianapolis, IN 46280, tghorner@hughes.net) and Michael A. Stokes (Waveform Commun., Indianapolis, IN)

Concussions impair cognitive function and muscle motor control; however, little is known about how this impairment affects speech production. In the present study, concussed athletes speech is recorded at the initial office visit and subsequent visits. The last recording, when the brain is determined to have recovered using present criteria, becomes the baseline. The vocabulary consists of seven h-vowel-d (hVd) words (who'd, heed, hood, hid, had, hud, and heard) produced three times each for a total of 21 productions. The study is focused on vowel characteristics and the limited coarticulatory effects of the hVd vocabulary make it ideal for the study. Duration measurements are made by experimenter analysis and the formant measurements are made using the automatic speech recognition engine ELBOW. The preliminary comparisons from the subjects completing the protocol show formant drift for three or more of the seven vowels, and duration is affected for each talker and each vowel. These results were anticipated since the impairment would affect articulatory movement and timing. The results as well as a discussion of development of an automated real-time concussion identification application will be presented.

5aSC27. Thai phonetically balanced word recognition test: Reliability evaluations and bias and error analysis. Adirek Munthuli (Elec. and Comput. Eng., Thammasat Univ., Khlong Luang, Pathumthani, Thailand), Chutamanee Onsuwan (Linguist, Thammasat Univ., Dept. of Linguist, Faculty of Liberal Arts, Thammasat University, Khlong Luang, Pathumthani 12120, Thailand, consuwan@hotmail.com), Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., Khlong Luang, Pathumthani, Thailand), and Krit Kosawat (Thailand National Electronics and Comput. Technol. Ctr., Khlong Luang, Pathumthani, Thailand)

Word recognition score (WRS) is one of the measuring techniques used in speech audiometry, a part of a routine audiological examination. The test's accuracy is crucial and largely depends on the test materials. With emphasis on phonetic balance, test-retest reliability, inter-list equivalency, and symmetrical phoneme occurrence, Thammasat University Phonetically Balanced Word Lists 2014 (TU PB'14) were created with five different lists, each with 25 Thai monosyllabic words. TU PB'14 reflects Thai phoneme distribution based on large-scale written Thai corpora, InterBEST [1]. To evaluate its validity and test-retest reliability, the lists were given at five intensity levels (15–55 dB HL) in test and retest sessions to 30 normal-hearing subjects. The differences in performance between the two sessions are not significantly large and correlation coefficients at the linear regions are all positive. Analysis of listeners' errors, including sequence recurrences, was carried out. Errors occurred predominantly in the case of initials, followed by finals and lexical tones. Confusion patterns of initials, finals, and tones are in line with those found for Thai speech sounds in noise condition. Interestingly, vowels are found to be most resistant to confusion. Finally, possible effect of lexical frequency is examined and discussed.

5aSC28. Talker variability in spoken word recognition: Evidence from repetition priming. Yu Zhang and Chao-Yang Lee (Ohio Univ., W239 Grover Ctr., Ohio University, Athens, OH 45701, yz137808@ohio.edu)

The effect of talker variability on the processing of spoken words is investigated using short-term repetition priming experiments. Prime-target pairs, either repeated (e.g., queen-queen) or unrelated (e.g., bell-queen), were produced by the same or different male speakers. Two interstimulus intervals (ISI, 50 and 250 ms) were used to explore the time course of repetition priming and voice specificity effects. The auditory stimuli were presented to 40 listeners, who completed a lexical decision task followed by a talker voice discrimination task. Results from the lexical decision task showed that the magnitude of priming was attenuated in the different-talker condition, indicating a talker variability effect on spoken word recognition. In contrast, the talker variability effect on priming did not differ between the two ISIs. Talker voice discrimination was faster and more accurate for nonword targets, but not for word targets, indicating a lexical status effect on voice discrimination. Taken together, these results suggest that talker variability affects recognition of spoken words, and that the effect cannot be simply attributed to non-lexical voice discrimination.

FRIDAY MORNING, 31 OCTOBER 2014 INDIANA F, 8:00 A.M. TO 12:30 P.M.

Session 5aUW

Underwater Acoustics: Acoustics, Ocean Dynamics, and Geology of Canyons

John A. Colosi, Cochair
Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943

James Lynch, Cochair
Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543

Chair's Introduction—8:00

Invited Papers

8:05

5aUW1. What do we know and what do we need to know about submarine canyons for acoustics? James Lynch, Ying-Tsong Lin, Timothy Duda, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu), and Glen Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA)

Acoustic propagation and scattering in marine canyons is an inherently 3-D problem, both for the environmental input (bottom topography and geology, biology, and physical oceanography) and the acoustic field. In this talk, we broadly examine what our knowledge is of these environmental fields, and what their salient effects should be upon acoustics. Examples from recent experiments off the United States and Taiwan will be presented, along with other historical data. Three dimensional acoustic modeling results will also be presented. Directions for future research will be discussed.

8:25

5aUW2. Ocean dynamics and numerical modeling of canyons and shelfbreaks. Pierre F. Lermusiaux (MechE, MIT, 77 Mass Ave., Cambridge, MA 02139, pierrel@mit.edu, Patrick Haley, Chris Mirabito (MIT, Cambridge, MA 02139), Timothy Duda, and Glen Gawarkiewicz (WHOI, Woods Hole, MA)

Multiscale ocean dynamics and multi-resolution numerical modeling of canyons and shelfbreaks are outlined. The dynamics focus is on fronts, currents, tides, and internal tides/waves that occur in these regions. Due to the topographic gradients and strong internal field gradients, nonlinear terms and non-hydrostatic dynamics can be significant. Computationally, a challenge is to achieve accurate simulations that resolve strong gradients over dynamically significant space- and time-scales. To do so, one component are high-order schemes that are more accurate for the same efficiency than lower-order schemes. A second is multi-resolution grids that allow optimized refinements, such as reducing errors near steep topography. A third are methods that allow to solve for multiple dynamics, e.g., hydrostatic and non-hydrostatic, seamlessly. To address these components, new hybridizable discontinuous Galerkin (HDG) finite-element schemes for (non)-hydrostatic physics including a nonlinear free-surface are introduced. The results of data-assimilative multi-resolution simulations are then discussed, using the primitive-equation MSEAS system and telescoping implicitly two-way nested domains. They correspond to collaborative experiments: (i) Shallow Water 06 (SW06) and the Integrated Ocean Dynamics and Acoustics (IODA) research in the Middle Atlantic Bight region; (ii) Quantifying, Predicting and Exploiting Uncertainty (QPE) in the Taiwan-Kuroshio region; and (iii) Philippines Straits Dynamics Experiment (PhilEx).

8:45

5aUW3. Internal tides in canyons and their effect on acoustics. Timothy F. Duda, Weifeng G. Zhang, Ying-Tsong Lin (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), and Aurelien Ponte (Laboratoire de Physique des Océans, IFREMER-CNRS-IRD-UBO, Plouzane, France)

Internal gravity waves of tidal frequency are generated as the ocean tides push water upward onto the continental shelf. Such waves also arrive at the continental slope from deep water and are heavily modified by the change in water depth. The wave generation and wave shoaling effects have an additional level of complexity where a canyon is sliced into the continental slope. Recently, steps have been taken to simulate internal tides in canyons, to understand the physical processes of internal tides in canyons, and also to compute the ramifications on sound propagation in and near the canyons. Internal tides generated in canyons can exhibit directionality, with the directionality being consistent with an interesting multiple-scattering effect. The directionality imparts a pattern to the sound-speed anomaly field affecting propagation. The directionality also means that short nonlinear internal waves, which have specific strong effects on sound, can have interesting patterns near the canyons. In addition to the directionality of internal tides radiated from canyons, the internal tide energy within the canyons can be patchy and may unevenly affect sound.

9:05

5aUW4. An overview of internal wave observations and theory associated with canyons and slopes. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

Topographic environments such as canyons and slopes are known to be regions of complex internal-wave behavior associated with wave generation, propagation, and dissipation. Much of this anomalous behavior stems from the kinematic constraint that internal waves must maintain their angle of propagation with respect to the horizontal even after interaction with a sloping boundary. In canyons or on slopes, waves propagating in from deep water or generated locally (mostly by tidal flows) either reflect back out to sea or intensify in energy density as they propagate up slope. In particular, wave intensification can lead to nonlinear phenomena including steepening, breaking, and dissipation. This talk will provide and overview of internal wave observations, modeling, and theory in canyons and on slopes with a particular emphasis on acoustically relevant aspects of the wave field.

9:25

5aUW5. Fiery ice from the sea: Marine gas hydrates. Ross Chapman (Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

Marine gas hydrates are cage-like structures of water containing methane or some higher hydrocarbons that are stable under conditions of high pressures and low temperatures. The hydrate structures are formed in sediments of continental margins and are found worldwide. The stability zone extends to about 200 m beneath the sea floor, and hydrates exist in several different forms within the zone, from massive ice-like features at cold seeps on the sea floor to finely distributed deposits in sediment pores over extensive areas. The base of the stability zone is characterized by a strong acoustic impedance change from high velocity hydrated sediments above to low velocity gas below. This acoustic feature generates a strong signal in seismic surveys called the Bottom Simulating Reflector, and it is widely used as an indicator of the presence of hydrates. This paper reviews the current knowledge of hydrate systems from research carried out on the Cascadia Margin off the west coast of Vancouver Island, and in the Gulf of Mexico. The hydrate distributions are different in each of these areas, leading to different effects in acoustic reflectivity.

9:45

5aUW6. South China Sea upper-slope sand dunes acoustics experiment. Ching-Sang Chiu, Ben Reeder (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyder Rd., Rm. 328, Monterey, CA 93943-5193, chiu@nps.edu), Linus Chiu (National Sun Yat-sen Univ., Kaohsiung, Taiwan), Yiing Jang Yang, and Chifang Chen (National Taiwan Univ., Taipei, Taiwan)

Very large subaqueous sand dunes were discovered on the upper continental slope of the Northeastern South China Sea. The spatial distribution and scales of these large sand dunes were mapped by two multibeam echo sounding (MBES) surveys, one in 2012 and the other in 2013. These two surveys represented two pilot cruises as part of a multiyear, US-Taiwan collaborative field study designed to characterize these sand dunes, the associated physical processes and the associated acoustic scattering physics. The main experiment will be carried out in 2014. The combination of MBES, coring and acoustic transmission data obtained from the two pilot cruises have

provided vital initial knowledge of (1) the spatial and temporal scales of the sand dunes from objective analysis, (2) the geoacoustic properties of the dunes based on forward modeling to matching the measured levels, and (3) the anisotropy and translational variability of the transmission loss based on a signal energy analysis of the repeated 1–2 kHz and 4–6 kHz FM signals transmitted by a calibrated sound source towed along two circular tracks, each surrounding a receiver. The results from the pilot cruises are presented and discussed. [The research is sponsored by the US ONR and the Taiwan NSC.]

10:05–10:20 Break

10:20

5aUW7. Three dimensional underwater acoustic modeling on continental slopes and submarine canyons. Ying-Tsong Lin, David Barclay, Timothy F. Duda, and Weifeng Gordon Zhang (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

Underwater sound propagation on slopes and canyons is influenced jointly and strongly by the complexity of topographic variability and ocean dynamics. Some integrated ocean and acoustic models have been developed and implemented to investigate such joint acoustic effects. In this talk, an integrated numerical model employing a time-stepping three-dimensional (3D) parabolic-equation (PE) acoustic modeling method and the Regional Ocean Modeling System (ROMS) is presented. Numerical examples of sound propagation and ambient noise in Mid-Atlantic Bight area with realistic environmental conditions are demonstrated. The sound propagation model reveals the focusing of sound due to concave canyon seafloor and the different level of temporal variability of focused and unfocused sound. The ambient noise model is constructed for surface wind generated noise, and the model shows the azimuthal dependency of noise field and its spatial coherence structure. Lastly, a simple sonar performance prediction is made to investigate the variability of the probability of detection in these complex underwater environments. [Work supported by the ONR.]

10:40

5aUW8. Three-dimensional effects in the sound propagation in area of coastal slope. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru) and Andrey Malykhin (Phys., Voronezh Univ., Voronezh, Russian Federation)

Coastal slope (wedge) in the ocean is well known “canonical” problem for analysis of manifestation of the horizontal refraction (3d effects) in a shelf zone. In given paper the following effects are reviewed:(1) Spatial variability of the sound field in given area. Areas of one-path and multipath propagation, shadow zones, and caustics in horizontal plane, their properties in dependence on the frequency, influence of bottom parameters;(2) Interference structure of the sound field in the horizontal plane in dependence on mode number and frequency;(3) Distribution of the sound field in vicinity of curvilinear coastal line, for example, gulf, bay, peninsula etc. (shadow zones, multipath area, and whispering gallery waveguide in horizontal plane);(4) Temporal variability of signals due to frequency dependence of the horizontal refraction and in turn pulse compression/decompression and time reversal in multipath area;(5) time-frequency diagrams; Mentioned and other effects can change properties of bottom and surface reverberation, scattering, noise field distribution, attenuation in area of coastal wedge. The corresponding estimations are presented. [Work was supported by BSF, Grant 2010471, RFBR-NSFC Grant 14-05-91180.]

Contributed Papers

11:00

5aUW9. Analytic prediction of acoustic coherence time scales in continental-shelf environments with random internal waves. Zheng Gong, Tianrun Chen (Mech. Eng., Massachusetts Inst. of Technol., 5-435, 77 Massachusetts Ave., Cambridge, MA 02139, zgong@mit.edu), Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), and Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

An analytical model derived from normal mode theory for the accumulated effects of range-dependent multiple forward scattering is applied to estimate the temporal coherence of the acoustic field forward propagated through a continental-shelf waveguide containing random three-dimensional internal waves. The modeled coherence time scale of narrow band low-frequency acoustic field fluctuations after propagating through a continental-shelf waveguide is shown to decay with a power-law of range to the -1/2 beyond roughly 1 km, decrease with increasing internal wave energy, to be consistent with measured acoustic coherence time scales. The model should provide a useful prediction of the acoustic coherence time scale as a function of internal wave energy in continental-shelf environments. The acoustic coherence time scale is an important parameter in remote sensing applications because it determines (i) the time window within which standard coherent processing such as matched filtering may be conducted, and (ii) the number of statistically independent fluctuations in a given measurement period that determines the variance reduction possible by stationary averaging.

11:15

5aUW10. Modeling three dimensional environment and broadband acoustic propagation in Arctic shelf-basin region. Mohsen Badiey, Andreas Muenchow, Lin Wan (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu), Megan S. Ballard (Appl. Res. Labs., Univ. of Texas, Austin, Delaware), David P. Knobles, and Jason D. Sagers (Appl. Res. Labs., Univ. of Texas, Austin, TX)

Rapid climate change over the last decade has created a renewed interest in the nature of underwater sound propagation in the Arctic Ocean. Changes in the oceanography and surface boundary conditions are expected to cause measurable changes in the propagation and scattering of low frequency sound. Recent measurements of a high-resolution three-dimensional (3-D) sound speed structure in a 50 x 50 km^2 region in an open-water shelf-basin region of the Beaufort Sea offer a unique and rare opportunity to study the effects of a complex oceanography on the acoustic field as it propagates from the deep basin onto the continental shelf. The raw oceanography data were analyzed and processed to create a 3-D sound speed field for the water column in the basin-slope-shelf area. Recent advances in both 2-D and 3-D acoustic modeling capability allow one to study the effects of the range- and azimuth-dependent water column layers on the frequency-dependent acoustic modal structure. Of particular interest is the nature of the 3-D and mode-coupling effects on the frequency response induced by the oceanography. The results will likely be useful in designing acoustic experiments with serious logistical constraints in the rapidly changing Arctic Ocean.

11:30

5aUW11. Underwater jet noise simulation based on a Large Eddy Simulation/Lighthill hybrid method. GuoQing Liu (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan 430074, China China, liugq_2010@163.com), Tao Zhang, YongOu Zhang (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei Province, China), Huajiang Ouyang (School of Eng., Univ. of Liverpool, Liverpool, United Kingdom), and Xu Li (School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, China)

In recent years, extensive researches about the numerical method for aeroacoustics noise simulation have been made. However, the research of hydrodynamic noise develops slowly. In this paper, a hybrid method of combining Large Eddy Simulation (LES) and Lighthill's acoustic analogy theory is established to compute the hydrodynamic noise, which is based on the preliminary study of the method for aerodynamic noise prediction under low Mach number. First, the model of three-dimensional underwater jet is determined by an experimental model. Meanwhile, the CFD mesh and the acoustic mesh are both prepared. Then, the flow field of underwater jet is simulated with LES. The characteristics of turbulent flow are analyzed by the pressure difference and the uniformity coefficient of velocity. After that, the noise of underwater jet is simulated using the theory of Lighthill's acoustic analogy. Finally, the solutions obtained by the hybrid method are compared with the experimental data available in open literature. In conclusion, the sound pressure level at the observation point agrees well with the experimental data. The LES/Lighthill hybrid method is able to compute the underwater jet noise and the hydrodynamic noise.

11:45

5aUW12. Formation sparse aperture antenna arrays based on the sequence Costas. Igor I. Anikin (Concern CSRI Elektropribor, JSC, 30, Malaya Posadskaya Ul., St. Petersburg 197045, Russian Federation, anikin1952@bk.ru)

To obtain high spatial resolution in sonar, ultrasonic image, radar, seismic, and radio astronomy use active antenna arrays that contain a large number of elements. To reduce the cost of such an antenna arrays used with sparse aperture. In this approach, the antenna array is partitioned into several subarrays. Geometric size subarray equivalent equidistant placement Nc * Nc elements. Subarrays filled Nc elements arranged according to the sequence Costas Nc-th order. Each filled subarrays own Costas sequence. As a result, the number of elements in the array is reduced in times Nc. Form the beam pattern in the main sections close to the plane shape of the beam pattern of equidistant antenna array, and the directivity factor is almost independent of frequency band. The upper frequency is reduced in the directivity factor $(\pi \cdot Nc)/2$ times as compared with the plane equidistant antenna array. Using a decaying distribution can be reduced in amplitude level of the side lobe in the principal planes. Thus, by setting the order of the Costas sequence can in each case to optimize the degree of reduction of the number of elements in the array antenna at a predetermined directivity factor.

12:00

5aUW13. Ultra low frequency electromagnetic underwater sound source. Wei Lu and Yu Lan (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St.,Nangang District, Harbin 150001, China, luwei@hrbeu.edu.cn)

A detail analysis is presented of one ultra low frequency sound source which is smaller and lighter than conventional piezoelectric ultra low frequency sound source. This sound source is single piston vibration using the electromagnetic principle. The radiation characteristic of sound source is researched by single piston radiation model in low frequency. The dynamic characteristic such as resonant frequency, vibrant displacement of sound source is researched by analytic method and finite element method. In analytic method, the electricity-magnetism, magnetism-force, and force-vibration conversion models of sound source are established by differential equations in different coupling physical fields, and the dynamic characteristics based on the conversion models are simulated by combining and solving differential equations using the MATLAB/SIMULINK. In finite element method, using transient solver of electromagnetic analysis finite element software Ansoft, the dynamic characteristics of sound source are solved. Optimizing the dynamic characteristic of sound source by adjusting magnetic circuit, drive coil, and elastic component parameters, the resonant frequency and radiated sound power of sound source are determined. One prototype sound source design, in which the source level is 184 dB in frequency 73 Hz by calibration, is fabricated that demonstrated proof-of-concept.

12:15

5aUW14. Simplex underwater acoustic communications using passive time reversal. Lin Sun, Haisen Li, Bo Zou, and Ruo Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin Eng. University, No.145 Nantong St., Nangang District, Harbin City, Heilongjiang Province., Harbin 150001, China, sunlinhrb@sina.com)

The spatial-temporal compression, which is achieved through using simple time reversal (TR) process, can reduce the inter-symbol interference and increase the signal strength. The active TR needs two-way propagation, so it cannot be used in simplex underwater acoustic communications. Based on the one-way propagation property of passive TR, a simplex underwater acoustic communication method using passive TR is proposed. The proposed method is considered in two scenarios: uplink transmission from a single send-only element to an array and downlink transmission from an array to a single receive-only element. The principle of proposed method is analyzed in theory and the performance of proposed method is verified through experiment. Results demonstrate that passive TR process can improve the output signal-to-noise ratio and decrease the bit error rate, so the performance of proposed method is superior to that of simplex acoustic communication method without using passive TR.

This document is frequently updated; the current version can be found online at the Internet site: <http://scitation.aip.org/content/asa/journal/jasa/info/authors>.

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, and an optional 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.

PACS numbers: 43.05.Gv

TABLE OF CONTENTS

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

a)E-mail: jasa@aip.org

features that are desired for the submitted manuscript. This document extracts many of the style suggestions found in the *AIP Style Manual,*[1] which is available online at the internet site <http://www.aip.org/pubservs/style/4thed/toc.html>. The *AIP Style Manual,* although now somewhat dated and not specifically directed toward publication in the *Journal of the Acoustical Society of America* (JASA), is a substantially more comprehensive document, and authors must make use of it also when preparing manuscripts. If conflicting instructions are found in the two documents, those given here take precedence. (Authors should also look at recent issues of the *Journal* for examples of how specific style issues are handled.) Conscientious consideration of the instructions and advice given in the two documents should considerably increase the likelihood that a submitted manuscript will be rapidly processed and accepted for publication.

II. ONLINE HANDLING OF MANUSCRIPTS

All new manuscripts intended for possible publication in the *Journal of the Acoustical Society of America* should be submitted by an online procedure. 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 Peer X-Press (PXP) system. The Acoustical Society of America contracts with AIP Publishing LLC 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 PXP system, and the undertaking of separate actions, such as the submission of a manuscript, requires that one first log-in to the system at http://jasa.peerx-press.org/cgi-bin/main.plex.

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 data base, so if you are a member, you in principle may already have a user name and password, but you will have to find out what they are. On the login page, you click on the item "Unknown/Forgotten Password." On the new page that comes up after you do this, give your first name and last name. After you have filled in this information, just click on "mailit." You will then get a e-mail message with the subject line "FORGOTTEN PASSWORD." The system will actually give you a new password if you had ever used the system before. After you get this new password, you can change it to something easy to remember after you login.

Once you have your "user name" and "password" you go to the log-in page again, and give this information when you log-in. You will first be asked to change your password. After you do this, a "task page" will appear. At the bottom of the page there will be an item *Modify Profile/Password.* Click on this. Then a Page will appear with the heading *Will you please take a minute to update the profile?*

If you are satisfied with your profile and password, then you go to the top of the Task page and click on the item *Submit Manuscript* that appears under *Author Tasks.* Then you will see a page titled *Manuscript Submission Instructions.* Read what is there and then click *continue* at the bottom of the page.

B. Overview of the editorial process

(1) An author denoted as the corresponding author submits a manuscript for publication in the *Journal.*
(2) One of the *Journal's* Associate Editors is recruited to handle the peer-review process for the manuscript.
(3) The Associate Editor recruits reviewers for the manuscript via the online system.
(4) The reviewers critique the manuscript, and submit their comments online via the Peer X-Press system.
(5) 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.
(6) 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.

Another document that you should first read is the document *Transfer of Copyright Agreement,* which is downloadable from the same site. When you submit your manuscript online you will be asked to certify that you and your coauthors agree to the terms set forth in that document. What is in that document has been carefully worded with extensive legal advice and which has been arrived at after extensive discussion within the various relevant administrative committees of the Acoustical Society of America. It is regarded as a very liberal document in terms of the rights that are allowed to the authors. One should also note the clause: The author(s) agree that, insofar as they are permitted to transfer copyright, all copies of the article or abstract shall include a copyright notice in the ASA's name. (The word "permitted" means permitted by law at the time of the submission.) The terms of the copyright agreement are non-negotiable. The Acoustical Society does not have the resources or legal assistance to negotiate for exceptions for individual papers, so please do not ask for such special considerations. Please read the document carefully and decide whether you can provide an electronic signature (clicking on an appropriate check box) to this agreement. If you do not believe that you can in good conscience give such an electronic signature, then you should not submit your manuscript.

Given that one has met the ethical criteria and agreed to the terms of the copyright transfer agreement, and that one has decided to submit a manuscript, one should first gather together the various items of information that will be requested during the process, and also gather together various files that one will have to upload.

Information that will be entered into the PeerX-Press submission form and files to be uploaded include:

(1) Data for each of the authors:
  (a) First name, middle initial, and last name
  (b) E-mail address
  (c) Work telephone number
  (d) Work fax number
  (e) Postal address (required for corresponding author, otherwise optional)

(2) Title and running title of the paper. The running title is used as the footline on each page of the article. (The title is limited to 17 words and the running title is limited to six words and up to 50 characters and spaces; neither may include any 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) Principal ASA-PACS number that characterizes the subject matter of the paper and which will be used to determine the section of the *Journal* in which the published paper will be placed. Note that if the PACS number you list first is too generic, e.g., 43.20, that may result in a delay in processing your paper.

(5) A short prioritized list of Associate Editors suggested for the handling of the manuscript.

(6) Contact information (name, e-mail address, and institution) of suggested reviewers (if any), and/or names of reviewers to exclude and reasons why.

(7) Cover letter file (optional, with some exceptions). Material that would ordinarily have been in the cover letter is now supplied by answering online questions and by filling out the online form. However, if an author needs to supply additional information that should be brought to the attention of the editor(s) and/or reviewer(s), a cover letter should be written and put into the form of an electronic file.

(8) Properly prepared manuscript/article file in LaTeX, Word, or WordPerfect format. (The requirements for a properly prepared manuscript are given further below.) It is also possible to submit your file in PDF but this is not desirable since the entire manuscript must be retyped. 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 insure that the submitted manuscript/article file is of reasonable length, no more than 2 MB.

(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 will appear as a list in the manuscript itself.) The figures should not have the figure numbers included on the figures in the files as such, and it is the responsibility of the corresponding author to see that the files are uploaded in proper order. Authors may upload figures in a zip file (figure files must be numbered in order using 1, 2, etc. If figures have parts they must be numbered 1a, 1b, 1c, etc.). [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 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 Peer X-Press *Task Page* and can select the option of submitting a new manuscript. The resulting process leads the corresponding author through a sequence of screens.

The first screen will display a series of tabs including: Files, Manuscript Information, Confirm Manuscript, and Submit. Clicking on these tabs displays the tasks that must be completed for each step in the submission. Red arrows denote steps that have not been completed. Green arrows are displayed for each tab where the step has been successfully completed.

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. They also check on the estimated length of the manuscript in the event that the author indicates that page charges will not be paid. If all is in order, the Manuscript Coordinator initiates the process, using the ASA-PACS numbers 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 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 they 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 $1,080, although there would be no mandatory charge if the length were 12 pages.)

To estimate the extent of the page charges, count 3 manuscript pages (double-spaced lines, with wide margins) as equivalent to one printed page, and count 4 figures or tables as equivalent to one printed page. If this number exceeds 12 and your institution and/or sponsor will not pay the page charges, please shorten your paper before submitting it.

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 figures that contain parts are submitted in separate files for each part, the $325 charge applies to each 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.

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. 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 submitted by the online procedure 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 insure a uniform style for publications in the *Journal* and partly to insure that the copy-editing process will be maximally effective in producing a quality publication. For the latter reason, adequate white space throughout the manuscript is desired to allow room for editorial corrections, which will generally be hand-written on a printed hard-copy. 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 or unless they are defined in the title.
(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 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).
(13) Line numbers to assist reviewers in commenting on the manuscript may be included but they are not mandatory.

B. 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 0.85 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.) Italics are similarly designated with a single straight underline in black pencil. It is preferred that vectors be designated by bold face 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.

C. 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, with one complete affiliation for each author appearing immediately after the author's name, 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. (This includes the abstract with a separate line giving a prioritized listing of the ASA-PACS numbers that apply to the manuscript. The selected PACS numbers should be taken only from the appendix concerned with acoustics of the overall PACS listing.) 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 ordinarily not be included in the "Article" file. Authors do, however, have the option of including figures embedded in the text, providing there is no ambiguity in distinguishing figure captions from the manuscript text proper. This is understood to be done only for the convenience of the reviewers. Such embedded figures will be ignored in the production editing process. The figures that will be used are those that were uploaded, one by one as separate files, during the online submission process.

D. 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 will be printed at the bottom of each printed page, other than the first, when the paper appears in the *Journal.* Because the printing of running titles follows an abbreviated identification of the authors, the maximum permissible length depends critically on the number of the authors and the lengths of their names. The running title also 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. (If authors believe that the inclusion of a less common acronym in the title will help in information retrieval and/or will help some readers to better understand what is the subject matter of the paper, then that acronym should be explicitly defined in the title.) Ideally, titles should be such that one can easily identify the principal ASA-PACS numbers for the paper, and consequently they should contain appropriate key words. 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* does not allow 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 each author may be contacted by mail by interested readers; e-mail addresses are optional. 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 D.C. (for District of Columbia). If the address is in the United States, omit the country name.

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 lower case 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:[2]

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[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:

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[b]Author to whom correspondence should be addressed. Electronic mail: name@servername.com

E. 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). Displayed equations that are set apart from the text count as 40 words. 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.

On the same page of the abstract, but separated from the abstract by several blank lines, the authors must give the principal ASA-PACS number for the paper, followed by up to three other ASA-PACS numbers that apply. This should be in the format exemplified below:

PACS numbers: 43.30.Pc, 43.30.Sf

The principal ASA-PACS number must be the first in this list. All of the selected PACS numbers must begin with the number 43, this corresponding to the appendix of the overall PACS listing that is concerned with acoustics. Authors are requested not to adopt a principal PACS number in the category of General Linear Acoustics (one beginning with 43.20) unless there is no specific area of acoustics with which the subject matter can be associated. The more specific is the principal PACS number, the greater likelihood that an appropriate match may be made with an Associate Editor, and the greater likelihood that appropriate reviewers will be recruited. When the paper is printed, the list of ASA-PACS numbers will be immediately followed on the same line by the initials, enclosed in brackets, of the Associate Editor who handled the manuscript.

F. 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 numbered 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 numbered by capital letters: A, B, C, etc. The typing of first subheadings is bold-face, 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 bold-face, also with only the first word and proper nouns capitalized. Third subheadings appear in the text at the beginning of paragraphs. These are numbered by lower case letters (a, b, c, etc.) and these are typed in italics (not bold-faced). 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 numbered by upper-case 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 can 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,[3] 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. In particular, do not include references that merely state: "personal communication." (Possibly, one can give something analogous to this in a textual footnote, but only as a means of crediting an idea or pinpointing a source. In such a case an explanatory sentence or sentence fragment is preferred to the vague term of "personal communication.") 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 thesis, 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 multimedia (e.g., movie files, audio files, animated .gifs, 3D rendering files), color figures, data tables, and text (e.g., appendices) 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 coauthor.

Journal titles must ordinarily be abbreviated, and each abbreviation must be in a "standard" form. The *AIP Style Manual*[1] gives a lengthy list of standard abbreviations that are used for journals that report physics research, but the interdisciplinary nature of acoustics is such that the list omits many journals that are routinely cited in the *Journal of the Acoustical Society of America.* For determination of what abbreviations to use for journals not on the list, 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" (in Japanese), J. Acoust. Soc. Jpn. **56**, 143–147 (2000).

[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 4D. 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, p. 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" (in Russian), Akust. Zh. **32**, 129–131 (1986); English transl.: 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 Inst. (2002). ANSI S12.60 (R2009) American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools (American National Standards Inst., New York).

Ando, Y. **(1982)**. "Calculation of subjective preference in concert halls," J. Acoust. Soc. Am. Suppl. 1 **71,** S4-S5.

Bacon, S. P. **(2000)**. "Hot topics in psychological and physiological acoustics: Compression," J. Acoust. Soc. Am. **107,** 2864(A).

Bergeijk, W. A. van, Pierce, J. R., and David, E. E., Jr. **(1960).** *Waves and the Ear* (Doubleday, Garden City, NY), Chap. 5, pp. 104-143.

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** (part 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 ...," J. Acoust. Soc. Am. **107,** 2329-2337.

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 must not 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 insure 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 http://www.langsci.ucl.ac.uk/ipa/. The display of the most recent version of the alphabet can be found at http://www.langsci.ucl.ac.uk/ipa/ipachart.html.

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, PXP'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. Because of the advances in technology and the increasingly wider use of computers in desk-top publishing, it is strongly preferred that authors use computers exclusively in the preparation of illustrations. If any figures are initially in the form of hard copy, they should be scanned with a high quality scanner and converted to electronic form. Each figure that is to be included in the paper should be cast into one of several acceptable formats (TIFF, EPS, JPEG, or PS) and put into a separate file.

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. The numbering of the figures, insofar as the online submission process is concerned, is achieved by uploading the individual figure files in the appropriate sequence. The author should take care to make sure that the sequence is correct, but the author will also have the opportunity to view the merged manuscript and to check on this sequencing.

For the most part, figures must be designed so that they will fit within one column (3-3/8″) of the page, and yet be intelligible to the reader. In rare instances, figures requiring full page width are allowed, but the choice for using such a figure must not be capricious.

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. Various clichés, such as Alice's—"and what is the use of a book without pictures?"—are strongly applicable to journal articles in acoustics. The absence of any such figures in a manuscript, even though they might have improved the clarity of the paper, is often construed as an indication of a callous lack of sympathy for the reader's potential difficulties when attempting to understand a paper.

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 lower case 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. If, however, the textual footnote style is used and the reference is not given in the text itself, an explicit reference listing must be given in the table footnote itself. This should contain the bare minimum of information necessary for a reader to retrieve the reference. In general, it is recommended that no footnote refer to references that are not already cited in the text.

VI. THE COVER LETTER

The submission of an electronic file containing a cover letter is now optional. Most of the *Journal's* requirements previously met by the submission of a signed cover letter are now met during the detailed process of online submission. The fact that the manuscript was transmitted by the corresponding author who was duly logged onto the system is taken as prima facie proof that the de facto transmittal letter has been signed by the corresponding author.

There are, however, some circumstances where 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 Information for 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 Peer X-Press 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. Subject classification, ASA-PACS

Authors are asked in their online submittal and in their manuscript to identify the subject classification of their paper using the ASA-PACS system. The subject index of the *Journal* presently follows a specialized extension of the *Physics and Astronomy Classification Scheme*[4] (PACS) maintained by AIP Publishing. Numbers in this scheme pertaining to Acoustics have the general form: 43.nn.Aa, where n denotes a digit, A denotes a capital alphabetical letter, and a denotes a lower case letter. An amplified version of the section 43 listing appears as an appendix to AIP Publishing's document, and this is here referred to as the ASA-PACS system. The ASA-PACS listing for acoustics appears at the end of each volume of the *Journal* preceding the index (June and December issues). It can also be found by first going to the *Journal's* site <http://scitation.aip.org/content/asa/journal/jasa/info/authors> and then clicking the item: *Physics and Astronomy Classification Scheme (PACS), Section 43, Acoustics.* (On the CD distribution of the *Journal,* the appropriate file for the index of each volume is jasin.pdf. The listing of the ASA-PACS numbers is at the beginning of this file.) It is the authors' responsibility to identify a principal ASA-PACS number corresponding to the subject matter of the manuscript and also to identify all other ASA-PACS numbers (up to a total of four) that apply.

B. 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>. (On the CD distribution of the *Journal,* the appropriate file is jasae.pdf.)

Authors are not constrained to select Associate Editors specifically identified with their choice of principal ASA-PACS number and should note that the *Journal* has special Associate Editors for Mathematical Acoustics, Computational Acoustics, and Education in Acoustics. Review and tutorial articles are ordinarily invited; submission of unsolicited review 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.

C. 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.

2. Education in acoustics articles

Such papers should be of potential interest to acoustics educators. Examples include descriptions of laboratory experiments and of classroom demonstrations. Papers that describe computer simulations of basic acoustical phenomena also fall within this category. Tutorial discussions on how to present acoustical concepts, including mathematical derivations that might give students additional insight, are possible contributions.

3. 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.

4. 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.*

5. 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.

6. Replies to comments

Authors whose previously published paper has stimulated the submission of a comments paper, and which has subsequently been accepted, have the opportunity to reply to the comments. They are usually (but not invariably) notified of the acceptance of the comments paper, and the *Journal* prefers that the comments and the reply be published in successive pages of the same issue, although this is not always practicable. Replies are also reviewed using criteria similar to those of comments papers. As in the case of comments papers, the format of the title of a reply paper is rigidly prescribed, and examples can be found in the previous issues of the *Journal.*

7. 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 Peer X-Press 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.*

8. 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.

9. 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 PXP submittal process is used primarily to facilitate the incorporation of the reviews into the Journal.

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 and eloquence 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 insure 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 principal 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.* When such happens, it is often the case that the article either cites no previously published papers in the *Journal* or else cites no recent papers in any of the other journals that are commonly associated with acoustics. Given that the *Journal* has been in existence for over 80 years and has published of the order of 35,000 papers on a wide variety of acoustical topics over its lifetime, the absence of any references to previously published papers in the *Journal* raises a flag signaling the possibility that the paper lies outside the de facto scope of the *Journal.*

Authors concerned that their work may be construed by the Editors as not being within the scope of the *Journal* can strengthen their case by citing other papers published in the *Journal* that address related topics.

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[5] 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.*

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[6] 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.

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*[7]) 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.* Although now a de facto international journal, JASA because of its historical origins requires manuscripts to follow US English style conventions.

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,[8] Merriam-Webster's Collegiate Dictionary, 11th Edition,[9] Strunk and White s Elements of Style,[10] and the Chicago Manual of Style.[11] (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, and the instructions given here are intended that there be ample white space in the printed-out manuscript that such copy-editing can be carried out. 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,[12] or in a dictionary specifically devoted to acoustics such as the Dictionary of Acoustics[13] 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, recently 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 lower case, 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. An article containing, say, three or more acronyms in every paragraph will be regarded as pretentious and deliberately opaque.

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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[2] M. Mellody and G. H. Wakefield, "The time-frequency characteristics of violin vibrato: Modal distribution analysis and synthesis," J. Acoust. Soc. Am. **107,** 598-611 (2000).

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[4] American Institute of Physics, *Physics and Astronomy Classification Scheme 2003.* A paper copy is available from AIP Publishing LLC, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. It is also available online at the site <http://www.aip.org/pacs/index.html>.

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ASSOCIATE EDITORS IDENTIFIED WITH PACS CLASSIFICATION ITEMS

The Classification listed here is based on the Appendix to Section 43, “Acoustics,” of the current edition of the Physics and Astronomy Classification Scheme PACS of AIP Publishing LLC. The full and most current listing of PACS can be found at the internet site <http://www.aip.org/pubservs/pacs.html>. In the full PACS listing, all of the acoustics items are preceded by the primary classification number 43. The listing here omits the prefatory 43; a listing in the AIP Publishing document such as 43.10.Ce will appear here as 10.Ce.

The present version of the Classification scheme is intended as a guide to authors of manuscripts submitted to the Journal who are asked at the time of submission to suggest an Associate Editor who might handle the processing of their manuscript. Authors should note that they can also have their manuscripts processed from any of the special standpoints of (i) Applied Acoustics, (ii) Computational Acoustics, (iii) Mathematical Acoustics, or (iv) Education in Acoustics, and that there are special Associate Editors who have the responsibility for processing manuscripts from each of these standpoints.

The initials which appear in brackets following most of the listings correspond to the names of persons on the Editorial Board i.e., Associate Editors who customarily edit material that falls within that classification. A listing of full names and institutional affiliations of members of the Editorial Board can be found on the back cover of each issue of the *Journal*. A more detailed listing can be found at the internet site <http://asadl.org/jasa/for_authors_jasa>. The most current version of the present document can also be found at that site.

[05]	**Acoustical Society of America**
05.Bp	Constitution and bylaws [EM]
05.Dr	History [ADP]
05.Ft	Honorary members [EM]
05.Gv	Publications ARLO. Echoes. ASA Web page, electronic archives and references [ADP]
05.Hw	Meetings [EM]
05.Ky	Members and membership lists, personal notes, fellows [EM]
05.Ma	Administrative committee activities [EM]
05.Nb	Technical committee activities; Technical Council [EM]
05.Pc	Prizes, medals, and other awards [EM]
05.Re	Regional chapters [EM]
05.Sf	Obituaries
[10]	**General**
10.Ce	Conferences, lectures, and announcements (not of the Acoustical Society of America) [EM]
10.Df	Other acoustical societies and their publications]; online journals and other electronic publications [ADP]
10.Eg	Biographical, historical, and personal notes (not of the Acoustical Society of America) [EM]
10.Gi	Editorials, Forum [ADP], [NX]
10.Hj	Books and book reviews [PLM]
10.Jk	Bibliographies [EM], [ADP]
10.Km	Patents [DLR], [SAF]
10.Ln	Surveys and tutorial papers relating to acoustics research, tutorial papers on applied acoustics [ADP], [NX]
10.Mq	Tutorial papers of historical and philosophical nature [ADP], [NX], [WA]
10.Nq	News with relevance to acoustics nonacoustical theories of interest to acoustics [EM], [ADP]
10.Pr	Information technology, internet, nonacoustical devices of interest to acoustics [ADP], [NX]
10.Qs	Notes relating to acoustics as a profession [ADP], [NX]
10.Sv	Education in acoustics, tutorial papers of interest to acoustics educators [LLT], [WA], [BEA], [VWS], [PSW]
10.Vx	Errata [ADP]
[15]	**Standards [SB], [PDS]**
[20]	**General linear acoustics**
20.Bi	Mathematical theory of wave propagation [MD], [SFW], [ANN], [RM], [RKS], [KML], [CAS]
20.Dk	Ray acoustics [JES], [SFW], [ANN], [JAC], [KML], [TFD], [TK]
20.El	Reflection, refraction, diffraction of acoustic waves [JES], [OU], [SFW], [RM], [KML], [GH], [TFD], [TK]
20.Fn	Scattering of acoustic waves [LLT], [JES], [OU], [SFW], [RM], [KML], [GH], [TK]
20.Gp	Reflection, refraction, diffraction, interference, and scattering of elastic and poroelastic waves [OU], [RM], [DF], [RKS], [JAT], [DSB], [GH]
20.Hq	Velocity and attenuation of acoustic waves [MD], [OU], [SFW], [TRH], [RAS], [NPC], [JAT], [GH]
20.Jr	Velocity and attenuation of elastic and poroelastic waves [ANN], [NPC], [RKS], [GH]
20.Ks	Standing waves, resonance, normal modes [LLT], [SFW], [RM], [JDM]
20.Mv	Waveguides, wave propagation in tubes and ducts [OU], [LH], [RK], [JBL]
20.Px	Transient radiation and scattering [LLT], [JES], [ANN], [MDV], [DDE]
20.Rz	Steady-state radiation from sources, impedance, radiation patterns, boundary element methods [SFW], [RM], [FCS]
20.Tb	Interaction of vibrating structures with surrounding medium [LLT], [RM], [FCS], [LH]
20.Wd	Analogies [JDM]
20.Ye	Measurement methods and instrumentation [SFW], [TRH], [JDM], [GH]
[25]	**Nonlinear acoustics**
25.Ba	Parameters of nonlinearity of the medium [MD], [OAS], [ROC]
25.Cb	Macrosonic propagation, finite amplitude sound; shock waves [OU], [MDV], [PBB], [OAS], [ROC]
25.Dc	Nonlinear acoustics of solids [MD], [ANN], [OAS]
25.Ed	Effect of nonlinearity on velocity and attenuation [MD], [OAS], [ROC]
25.Fe	Effect of nonlinearity on acoustic surface waves [MD], [MFH], [OAS]
25.Gf	Standing waves; resonance [OAS], [MFH]
25.Hg	Interaction of intense sound waves with noise [OAS], [PBB]
25.Jh	Reflection, refraction, interference, scattering, and diffraction of intense sound waves [OU], [MDV], [PBB]
25.Lj	Parametric arrays, interaction of sound with sound, virtual sources [TRH]
25.Nm	Acoustic streaming [JDM], [OAS], [LH]
25.Qp	Radiation pressure [ROC]
25.Rq	Solitons, chaos [MFH]
25.Ts	Nonlinear acoustical and dynamical systems [MFH], [ROC]
25.Uv	Acoustic levitation [MFH]
25.Vt	Intense sound sources [ROC], [TRH]
25.Yw	Nonlinear acoustics of bubbly liquids [TGL], [SWY]
25.Zx	Measurement methods and instrumentation for nonlinear acoustics [ROC]
[28]	**Aeroacoustics and atmospheric sound**
28.Bj	Mechanisms affecting sound propagation in air, sound speed in the air [DKW], [VEO], [KML]
28.Dm	Infrasound and acoustic-gravity waves [DKW], [PBB]
28.En	Interaction of sound with ground surfaces, ground cover and topography, acoustic impedance of outdoor surfaces [OU], [KVH], [VEO], [KML]
28.Fp	Outdoor sound propagation through a stationary atmosphere, meteorological factors [DKW], [KML], [TK]
28.Gq	Outdoor sound propagation and scattering in a turbulent atmosphere, and in non-uniform flow fields [VEO], [PBB], [KML]
28.Hr	Outdoor sound sources [JWP], [PBB], [TK]
28.Js	Numerical models for outdoor propagation [VEO], [NAG], [DKW]
28.Kt	Aerothermoacoustics and combustion acoustics [AH], [JWP], [LH]
28.Lv	Statistical characteristics of sound fields and propagation parameters [DKW], [VEO]
28.Mw	Shock and blast waves, sonic boom [VWS], [ROC], [PBB]
28.Py	Interaction of fluid motion and sound. Doppler effect and sound in flow ducts [JWP], [AH], [LH]
28.Ra	Generation of sound by fluid flow, aerodynamic sound, and turbulence, [JWP], [AH], [PBB], [TK], [LH]
28.Tc	Sound-in-air measurements, methods and instrumentation for location, navigation, altimetry, and sound ranging [JWP], [KVH], [DKW]
28.Vd	Measurement methods and instrumentation to determine or evaluate atmospheric parameters, winds, turbulence, temperatures, and pollutants in air [JWP], [DKW]
28.We	Measurement methods and instrumentation for remote sensing and for inverse problems [DKW]
[30]	**Underwater sound**
30.Bp	Normal mode propagation of sound in water [BTH], [AMT], [MS], [NPC], [TFD]
30.Cq	Ray propagation of sound in water [JES], [BTH], [JAC], [TFD]
30.Dr	Hybrid and asymptotic propagation theories, related experiments [BTH], [JAC], [TFD]
30.Es	Velocity, attenuation, refraction, and diffraction in water, Doppler effect [BTH], [DRD], [JAC], [TFD]
30.Ft	Volume scattering [BTH], [APL]
30.Gv	Backscattering, echoes, and reverberation in water due to combinations of boundaries [BTH], [APL]
30.Hw	Rough interface scattering [BTH], [JES], [APL]
30.Jx	Radiation from objects vibrating under water, acoustic and mechanical impedance [BTH], [DSB], [DF], [EGW], [DDE]
30.Ky	Structures and materials for absorbing sound in water; propagation in fluid-filled permeable material [BTH], [NPC], [FCS], [TRH]
30.Lz	Underwater applications of nonlinear acoustics; explosions [BTH], [NAG], [OAS], [SWY]

30.Ma Acoustics of sediments; ice covers, viscoelastic media; seismic underwater acoustics [BTH], [NAG], [MS], [DSB]

30.Nb Noise in water; generation mechanisms and characteristics of the field [BTH], [KGS], [MS], [JAC], [SWY]

30.Pc Ocean parameter estimation by acoustical methods; remote sensing; imaging, inversion, acoustic tomography [BTH], [KGS], [AMT], [MS], [JAC], [ZHM], [HCS], [SED], [TFD], [APL]

30.Qd Global scale acoustics; ocean basin thermometry, transbasin acoustics [BTH], [JAC]

30.Re Signal coherence or fluctuation to sound propagation/scattering in the ocean [BTH], [KGS], [HCS], [TFD]

30.Sf Acoustical detection of marine life; passive and active [BTH], [CF], [DKM], [AMT], [MS], [MCH], [APL]

30.Tg Navigational instruments using underwater sound [BTH], [HCS], [JAC]

30.Vh Active sonar systems [BTH], [JES], [TRH], [ZHM], [DDE]

30.Wi Passive sonar systems and algorithms, matched field processing in underwater acoustics [BTH], [KGS], [HCS], [AMT], [MS], [SED], [ZHM]

30.Xm Underwater measurement and calibration instrumentation and procedures [BTH], [JAC], [TRH], [DDE]

30.Yj Transducers and transducer arrays for underwater sound; transducer calibration [BTH], [TRH], [DDE]

30.Zk Experimental modeling [BTH], [JES], [MS], [TFD]

[35] Ultrasonics, quantum acoustics, and physical effects of sound

35.Ae Ultrasonic velocity, dispersion, scattering, diffraction, and attenuation in gases [VMK], [MRH], [AGP], [GH], [TK]

35.Bf Ultrasonic velocity, dispersion, scattering, diffraction, and attenuation in liquids, liquid crystals, suspensions, and emulsions [VMK], [MRH], [AGP], [DSB], [NAG], [JDM], [GH]

35.Cg Ultrasonic velocity, dispersion, scattering, diffraction, and attenuation in solids; elastic constants [VMK], [MRH], [AGP], [MD], [MFH], [JDM], [JAT], [RKS], [GH], [TK]

35.Dh Pretersonics (sound of frequency above 10 GHz); Brillouin scattering [VMK], [MRH], [AGP], [MFH], [RLW]

35.Ei Acoustic cavitation, vibration of gas bubbles in liquids [VMK], [MRH], [AGP], [TGL], [NAG], [DLM]

35.Fj Ultrasonic relaxation processes in gases, liquids, and solids [VMK], [MRH], [AGP], [NAG]

35.Gk Phonons in crystal lattices, quantum acoustics [VMK], [MRH], [AGP], [DF], [LPF], [JDM]

35.Hl Sonoluminescence [VMK], [MRH], [AGP], [NAG], [TGL]

35.Kp Plasma acoustics [VMK], [MRH], [AGP], [MFH], [JDM]

35.Lq Low-temperature acoustics, sound in liquid helium [VMK], [MRH], [AGP], [JDM]

35.Mr Acoustics of viscoelastic materials [VMK], [MRH], [AGP], [LLT], [MD], [OU], [FCS], [KVH], [GH]

35.Ns Acoustical properties of thin films [VMK], [MRH], [AGP], [ADP], [TK]

35.Pt Surface waves in solids and liquids [VMK], [MRH], [AGP], [MD], [ANN], [GH], [TK]

35.Rw Magnetoacoustic effect; oscillations and resonance [VMK], [MRH], [AGP], [DAB], [DF], [LPF]

35.Sx Acoustooptical effects, optoacoustics, acoustical visualization, acoustical microscopy, and acoustical holography [VMK], [MRH], [AGP], [JDM], [TK]

35.Ty Other physical effects of sound [VMK], [MRH], [AGP], [MFH], [NAG]

35.Ud Thermoacoustics, high temperature acoustics, photoacoustic effect [VMK], [MRH], [AGP], [JDM], [TB]

35.Vz Chemical effects of ultrasound [VMK], [MRH], [AGP], [TGL]

35.Wa Biological effects of ultrasound, ultrasonic tomography [VMK], [MRH], [AGP], [DLM], [MCH], [SWY]

35.Xd Nuclear acoustical resonance, acoustical magnetic resonance [VMK], [MRH], [AGP], [JDM]

35.Yb Ultrasonic instrumentation and measurement techniques [VMK], [MRH], [AGP], [ROC], [GH], [KAW], [TK]

35.Zc Use of ultrasonics in nondestructive testing, industrial processes, and industrial products [VMK], [MRH], [AGP], [MD], [JAT], [ANN], [BEA], [GH], [TK]

[38] Transduction; acoustical devices for the generation and reproduction of sound

38.Ar Transducing principles, materials, and structures: general [MS], [DAB], [TRH], [DDE]

38.Bs Electrostatic transducers [MS], [KG], [DAB], [TRH], [MRB], [DDE]

38.Ct Magnetostrictive transducers [DAB], [TRH], [DDE]

38.Dv Electromagnetic and electrodynamic transducers [MS], [DAB], [TRH], [DDE]

38.Ew Feedback transducers [MS]

38.Fx Piezoelectric and ferroelectric transducers [DAB], [KG], [TRH], [MRB], [DDE]

38.Gy Semiconductor transducers [MS], [MRB]

38.Hz Transducer arrays, acoustic interaction effects in arrays [DAB], [TRH], [MS], [BEA], [MRB], [DDE]

38.Ja Loudspeakers and horns, practical sound sources [MS], [MRB], [DDE]

38.Kb Microphones and their calibration [MS], [MRB]

38.Lc Amplifiers, attenuators, and audio controls [MS]

38.Md Sound recording and reproducing systems, general concepts [MAH], [MRB]

38.Ne Mechanical, optical, and photographic recording and reproducing systems [MS]

38.Pf Hydroacoustic and hydraulic transducers [DAB]

38.Qg Magnetic and electrostatic recording and reproducing systems [MS]

38.Rh Surface acoustic wave transducers [MS], [TK]

38.Si Telephones, earphones, sound power telephones, and intercommunication systems [MS]

38.Tj Public address systems, sound-reinforcement systems [ADP]

38.Vk Stereophonic reproduction [ADP], [MRB]

38.Wl Broadcasting (radio and television) [ADP]

38.Yn Impulse transducers [MS]

38.Zp Acoustooptic and photoacoustic transducers [DAB], [MS]

[40] Structural acoustics and vibration

40.At Experimental and theoretical studies of vibrating systems [AJH], [NJK], [EAM], [KML], [EGW], [DDE], [DF], [DAB], [FCS], [LC]

40.Cw Vibrations of strings, rods, and beams [AJH], [NJK], [EAM], [DDE], [EGW], [DAB], [LPF], [JAT], [LC], [BEA]

40.Dx Vibrations of membranes and plates [AJH], [NJK], [EAM], [LLT], [MD], [EGW], [DAB], [DF], [LPF], [LC], [JBL], [DDE]

40.Ey Vibrations of shells [AJH], [NJK], [EAM], [DAB], [DF], [LPF], [EGW], [LC], [DDE]

40.Fz Acoustic scattering by elastic structures [AJH], [NJK], [EAM], [LLT], [KML], [ANN], [DSB], [TK], [EGW], [DDE]

40.Ga Nonlinear vibration [AJH], [NJK], [EAM], [JAT]

40.Hb Random vibration [AJH], [NJK], [EAM]

40.Jc Shock and shock reduction and absorption [AJH], [NJK], [EAM], [OU]

40.Kd Impact and impact reduction, mechanical transients [AJH], [NJK], [EAM], [FCS]

40.Le Techniques for nondestructive evaluation and monitoring, acoustic emission [AJH], [NJK], [EAM], [JAT], [BEA], [TK]

40.Ng Effects of vibration and shock on biological systems, including man [AJH], [NJK], [EAM], [MCH]

40.Ph Seismology and geophysical prospecting; seismographs [AJH], [NJK], [EAM], [MFH], [RKS], [ANN]

40.Qi Effect of sound on structures, fatigue; spatial statistics of structural vibration [AJH], [NJK], [EAM], [JAT], [DDE]

40.Rj Radiation from vibrating structures into fluid media [AJH], [NJK], [EAM], [LLT], [KML], [FCS], [EGW], [LC], [LH], [DDE]

40.Sk Inverse problems in structural acoustics and vibration [AJH], [NJK], [EAM], [KML], [EGW], [LC], [DDE]

40.Tm Vibration isolators, attenuators, and dampers [AJH], [NJK], [EAM], [LC]

40.Vn Active vibration control [AJH], [NJK], [EAM], [BSC], [LC]

40.Yq Instrumentation and techniques for tests and measurement relating to shock and vibration, including vibration pickups, indicators, and generators, mechanical impedance [AJH], [NJK], [EAM], [LC]

[50] Noise: its effects and control

50.Ba Noisiness: rating methods and criteria [GB], [SF], [BSF]

50.Cb Noise spectra, determination of sound power [GB], [KVH]

50.Ed Noise generation [KVH], [RK]

50.Fe Noise masking systems [BSF]

50.Gf Noise control at source: redesign, application of absorptive materials and reactive elements, mufflers, noise silencers, noise barriers, and attenuators, etc. [OU], [SFW], [RK], [FCS], [AH], [LC], [JBL], [LH]

50.Hg Noise control at the ear [FCS], [BSF]

50.Jh Noise in buildings and general machinery noise [RK], [KVH], [KML]

50.Ki Active noise control [BSC], [LC]

50.Lj Transportation noise sources: air, road, rail, and marine vehicles [GB], [SFW], [SF], [JWP], [KVH], [KML]

50.Nm Aerodynamic and jet noise [SF], [JWP], [AH], [LH]

50.Pn Impulse noise and noise due to impact [GB], [KVH], [SF]

50.Qp Effects of noise on man and society [GB], [BSF], [SF]

50.Rq Environmental noise, measurement, analysis, statistical characteristics [GB], [BSF], [SF]

50.Sr Community noise, noise zoning, by-laws, and legislation [GB], [BSF], [SF]

50.Vt Topographical and meteorological factors in noise propagation [PBB], [VEO]

50.Yw Instrumentation and techniques for noise measurement and analysis [GB], [KVH], [RK]

[55] Architectural acoustics

55.Br Room acoustics: theory and experiment; reverberation, normal modes, diffusion, transient and steady-state response [NX], [MV], [JES], [FCS]

55.Cs Stationary response of rooms to noise; spatial statistics of room response; random testing [NX], [MV], [JES]

55.Dt Sound absorption in enclosures: theory and measurement; use of absorption in offices, commercial and domestic spaces [NX], [MV], [JES], [FCS]

55.Ev Sound absorption properties of materials: theory and measurement of sound absorption coefficients; acoustic impedance and admittance [NX], [MV], [OU], [FCS]

55.Fw Auditorium and enclosure design [NX], [MV], [JES], [NX]

55.Gx Studies of existing auditoria and enclosures [NX], [MV], [JES]

55.Hy Subjective effects in room acoustics, speech in rooms [NX], [MV], [JES], [MAH]

55.Jz Sound-reinforcement systems for rooms and enclosures [NX], [MV], [MAH]

55.Ka Computer simulation of acoustics in enclosures, modeling [NX], [LLT], [MV], [JES], [SFW], [NAG]

55.Lb Electrical simulation of reverberation [NX], [MV], [MAH]

55.Mc Room acoustics measuring instruments, computer measurement of room properties [NX], [MV], [JES]

55.Nd Reverberation room design: theory, applications to measurements of sound absorption, transmission loss, sound power [NX], [MV]

55.Pe Anechoic chamber design, wedges [NX], [ADP]

55.Rg Sound transmission through walls and through ducts: theory and measurement [NX], [LLT], [FCS], [LC], [BEA]

55.Ti Sound-isolating structures, values of transmission coefficients [NX], [LLT], [LC]

55.Vj Vibration-isolating supports in building acoustics [NX], [ADP]

55.Wk Damping of panels [NX], [LLT]

[58] Acoustical measurements and instrumentation

58.Bh Acoustic impedance measurement [DAB], [FCS]

58.Dj Sound velocity [DKW], [TB], [GH], [TK]

58.Fm Sound level meters, level recorders, sound pressure, particle velocity, and sound intensity measurements, meters, and controllers [MS], [DKW], [TB], [KAW]

58.Gn Acoustic impulse analyzers and measurements [ADP]

58.Hp Tuning forks, frequency standards; frequency measuring and recording instruments; time standards and chronographs [MS]

58.Jq Wave and tone synthesizers [MAH]

58.Kr Spectrum and frequency analyzers and filters; acoustical and electrical oscillographs; photoacoustic spectrometers; acoustical delay lines and resonators [ADP]

58.Ls Acoustical lenses and microscopes [ADP]

58.Mt Phase meters [ADP]

58.Pw Rayleigh disks [ADP]

58.Ry Distortion: frequency, nonlinear, phase, and transient; measurement of distortion [MS]

58.Ta Computers and computer programs in acoustics [FCS], [DSB], [VWS]

58.Vb Calibration of acoustical devices and systems [DAB]

58.Wc Electrical and mechanical oscillators [ADP]

[60] Acoustic signal processing

60.Ac Theory of acoustic signal processing [KGS], [MAH]

60.Bf Acoustic signal detection and classification, applications to control systems [JES], [MRB], [PJL], [ZHM], [MAH], [JAC]

60.Cg Statistical properties of signals and noise [KGS], [MAH], [TFD]

60.Dh Signal processing for communications: telephony and telemetry, sound pickup and reproduction, multimedia [MAH], [HCS], [MRB]

60.Ek Acoustic signal coding, morphology, and transformation [MAH]

60.Fg Acoustic array systems and processing, beam-forming [JES], [ZHM], [HCS], [AMT], [MRB], [BEA], [TFD]

60.Gk Space-time signal processing other than matched field processing [JES], [ZHM], [JAC], [MRB]

60.Hj Time-frequency signal processing, wavelets [KGS], [ZHM], [CAS], [PJL]

60.Jn Source localization and parameter estimation [JES], [KGS], [MAH], [ZHM], [MRB], [SED]

60.Kx Matched field processing [AIT], [AMT], [SED]

60.Lq Acoustic imaging, displays, pattern recognition, feature extraction [JES], [KGS], [BEA], [MRB]

60.Mn Adaptive processing [DKW], [MRB]

60.Np Acoustic signal processing techniques for neural nets and learning systems [MAH], [AMT]

60.Pt Signal processing techniques for acoustic inverse problems [ZHM], [MRB], [SED]

60.Qv Signal processing instrumentation, integrated systems, smart transducers, devices and architectures, displays and interfaces for acoustic systems [MAH], [MRB]

60.Rw Remote sensing methods, acoustic tomography [DKW], [JAC], [ZHM], [AMT]

60.Sx Acoustic holography [JDM], [OAS], [EGW], [MRB]

60.Tj Wave front reconstruction, acoustic time-reversal, and phase conjugation [OAS], [HCS], [EGW], [BEA], [MRB]

60.Uv Model-based signal processing [ZHM], [MRB], [PJL]

60.Vx Acoustic sensing and acquisition [MS], [DKW]

60.Wy Non-stationary signal analysis, non-linear systems, and higher order statistics [PJL]

[64] Physiological acoustics

64.Bt Models and theories of the auditory system [BLM], [ICB], [FCS], [CAS], [CA], [ELP]

64.Dw Anatomy of the cochlea and auditory nerve [BLM], [AMS], [ANP], [SFW], [RRF], [CAS], [CA]

64.Fy Anatomy of the auditory central nervous system [BLM], [AMS], [ANP], [RRF], [CAS], [CA]

64.Gz Biochemistry and pharmacology of the auditory system [BLM], [CAS], [CA]

64.Ha Acoustical properties of the outer ear; middle-ear mechanics and reflex [BLM], [FCS], [CAS], [CA], [ELP]

64.Jb Otoacoustic emissions [BLM], [MAH], [CAS], [CA], [ELP]

64.Kc Cochlear mechanics [BLM], [KG], [CAS], [CA], [ELP]

64.Ld Physiology of hair cells [BLM], [KG], [CAS], [CA], [ELP]

64.Me Effects of electrical stimulation, cochlear implant [BLM], [ICB], [CAS], [CA], [ELP]

64.Nf Cochlear electrophysiology [BLM], [ICB], [KG], [CAS], [CA], [ELP]

64.Pg Electrophysiology of the auditory nerve [BLM], [AMS], [ICB], [CAS], [CA], [ELP]

64.Qh Electrophysiology of the auditory central nervous system [BLM], [AMS], [ICB], [CAS], [ELP]

64.Ri Evoked responses to sounds [BLM], [ICB], [CAS], [CA], [ELP]

64.Sj Neural responses to speech [BLM], [ICB], [CAS], [ELP]

64.Tk Physiology of sound generation and detection by animals [BLM], [AMS], [MCH], [CAS]

64.Vm Physiology of the somatosensory system [BLM], [MCH]

64.Wn Effects of noise and trauma on the auditory system [BLM], [ICB], [CAS], [ELP]

64.Yp Instruments and methods [BLM], [KG], [MAH], [CAS]

[66] Psychological acoustics

66.Ba Models and theories of auditory processes [EB], [CAS], [ELP], [JFC]

66.Cb Loudness, absolute threshold [MAS], [ELP]

66.Dc Masking [VMR], [EAS], [FJG], [LRB], [EB], [ELP], [JFC]

66.Ed Auditory fatigue, temporary threshold shift [EAS], [MAS], [ELP], [EB]

66.Fe Discrimination: intensity and frequency [VMR], [FJG], [EB]

66.Gf Detection and discrimination of sound by animals [ADP]

66.Hg Pitch [ADP]

66.Jh Timbre, timbre in musical acoustics [DD]

66.Ki Subjective tones [JFC]

66.Lj Perceptual effects of sound [VMR], [VB], [DB], [EB], [JFC]

66.Mk Temporal and sequential aspects of hearing; auditory grouping in relation to music [EAS], [FJG], [DB], [EB], [DD]

66.Nm Phase effects [EB], [JFC]

66.Pn Binaural hearing [VB], [LRB], [EB], [ELP], [NAG], [JFC]

66.Qp Localization of sound sources [VB], [FJG], [LRB], [EB], [ELP], [JFC]

66.Rq Dichotic listening [FJG], [LRB], [EB], [DD], [ELP], [JFC]

66.Sr Deafness, audiometry, aging effects [DS], [FJG], [ICB], [MAS], [ELP], [JFC]

66.Ts Auditory prostheses, hearing aids [DB], [VB], [FJG], [ICB], [MAS], [JFC], [EB], [ELP]

66.Vt Hearing protection [FCS]

66.Wv Vibration and tactile senses [MCH]

66.Yw Instruments and methods related to hearing and its measurement [ADP]

[70] Speech production

70.Aj Anatomy and physiology of the vocal tract, speech aerodynamics, auditory kinetics [ZZ], [CYE], [CHS], [SSN], [LK]

70.Bk Models and theories of speech production [ZZ], [CYE], [CHS]

70.Dn Disordered speech [ZZ], [CYE], [LK][CHS], [DAB]

70.Ep Development of speech production [CYE], [DAB], [CHS], [ZZ], [LK]

70.Fq Acoustical correlates of phonetic segments and suprasegmental properties: stress, timing, and intonation [CYE], [SSN], [DAB], [CGC],

70.Gr Larynx anatomy and function; voice production characteristics [CYE], [CHS], [LK], [ZZ]

70.Jt Instrumentation and methodology for speech production research [DAB], [CHS], [LK], [ZZ]

70.Kv Cross-linguistics speech production and acoustics [DAB], [LK]

70.Mn Relations between speech production and perception [CYE], [DAB], [CHS], [CGC], [ZZ]

[71] Speech perception

71.An Models and theories of speech perception [TCB], [MSS], [ICB], [MAH], [CGC]

71.Bp Perception of voice and talker characteristics [TCB], [MSS], [CGC], [JHM], [MSV], [MAH]

71.Es Vowel and consonant perception; perception of words, sentences, and fluent speech [TCB], [MSS], [DB], [CGC], [MAH]

71.Ft Development of speech perception [TCB], [MSS], [CA], [MAH], [DB]

71.Gv Measures of speech perception (intelligibility and quality) [TCB], [MSS], [VB], [ICB], [CGC], [MAH], [MAS]

71.Hw Cross-language perception of speech [TCB], [MSS], [MAH], [CGC]

71.Ky Speech perception by the hearing impaired [TCB], [MSS], [DB], [VB], [FJG], [ICB], [EB]

71.Lz Speech perception by the aging [TCB], [MSS], [DB], [MAH]

71.Qr Neurophysiology of speech perception [TCB], [MSS], [ICB], [MAH]

71.Rt Sensory mechanisms in speech perception [TCB], [MSS], [ICB], [MAH], [DB]

71.Sy Spoken language processing by humans [TCB], [MSS], [DB], [MSV], [MAH], [CGC]

[72] Speech processing and communication systems

72.Ar Speech analysis and analysis techniques; parametric representation of speech [CYE], [SSN]

72.Bs Neural networks for speech recognition [CYE], [SSN]

72.Ct Acoustical methods for determining vocal tract shapes [CYE], [SSN], [ZZ]

72.Dv Speech-noise interaction [CYE], [SSN]

72.Fx Talker identification and adaptation algorithms [CYE], [SSN]

72.Gy Narrow, medium, and wideband speech coding [CYE], [SSN]

72.Ja Speech synthesis and synthesis techniques [CYE], [SSN], [SAF]

72.Kb Speech communication systems and dialog systems [CYE]

72.Lc Time and frequency alignment procedures for speech [CYE], [SSN]

72.Ne Automatic speech recognition systems [CYE], [SSN]

72.Pf Automatic talker recognition systems [CYE], [SSN]

72.Qr Auditory synthesis and recognition [CYE], [SSN]

72.Uv Forensic acoustics [CYE]

[75] Music and musical instruments

75.Bc Scales, intonation, vibrato, composition [DD], [MAH]

75.Cd Music perception and cognition [DD], [MAH], [DB]

75.De Bowed stringed instruments [TRM], [JW]

75.Ef Woodwinds [TRM], [JW], [AH]

75.Fg Brass instruments and other lip vibrated instruments [TRM], [JW], [ZZ]

75.Gh Plucked stringed instruments [TRM], [JW]

75.Hi Drums [TRM]

75.Kk Bells, gongs, cymbals, mallet percussion and similar instruments [TRM]

75.Lm Free reed instruments [TRM], [JW], [AH], [ZZ]

75.Mn Pianos and other struck stringed instruments [TRM]

75.Np Pipe organs [TRM], [JW]

75.Pq Reed woodwind instruments [AH], [TRM], [JW], [ZZ]

75.Qr Flutes and similar instruments [AH], [TRM], [JW]

75.Rs Singing [DD], [TRM], [JW]

75.St Musical performance, training, and analysis [DD], [DB]

75.Tv Electroacoustic and electronic instruments [DD]

75.Wx Electronic and computer music [MAH]

75.Xz Automatic music recognition, classification and information retrieval [DD], [SSN]

75.Yy Instrumentation measurement methods for musical acoustics [TRM], [JW]

75.Zz Analysis, synthesis, and processing of musical sounds [DD], [MAH]

[80] Bioacoustics

80.Cs Acoustical characteristics of biological media: molecular species, cellular level tissues [MLD], [RRF], [DLM], [TK], [SWY], [GH], [KAW], [TK]

80.Ev Acoustical measurement methods in biological systems and media [CCC], [DLM], [MLD], [RRF], [SWY], [GH], [KAW]

80.Gx Mechanisms of action of acoustic energy on biological systems: physical processes, sites of action [MLD], [ANP], [RRF], [GH], [SWY], [KAW]

80.Jz Use of acoustic energy (with or without other forms) in studies of structure and function of biological systems [MLD], [TJR], [ANP], [RRF], [DLM], [GH], [SWY], [KAW]

80.Ka Sound production by animals: mechanisms, characteristics, populations, biosonar [MLD], [WWA], [CT], [AMS], [ANP], [DKM], [JJF], [AMT], [ZZ]

80.Lb Sound reception by animals: anatomy, physiology, auditory capacities, processing [MLD], [AMS], [ANP], [DKM], [JJF]

80.Nd Effects of noise on animals and associated behavior, protective mechanisms [MLD], [AMS], [ANP], [DKM], [JJF], [AMT]

80.Pe Agroacoustics [RRF], [WA], [MCH]

80.Qf Medical diagnosis with acoustics [MDV], [DLM], [GH], [SWY], [KAW]

80.Sh Medical use of ultrasonics for tissue modification (permanent and temporary) [DLM], [ROC], [MDV], [GH], [SWY], [KAW]

80.Vj Acoustical medical instrumentation and measurement techniques [DLM], [MCH], [MDV], [GH], [SWY], [KAW]

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: asa@aip.org

Acentech Incorporated
www.acentech.com
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Consultants in Acoustics, Audiovisual and Vibration

ACO Pacific Inc.
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Producer of Aircraft and Aerospace Products

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D'Addario & Company, Inc.
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Farmingdale, New York
D'Addario strings for musical instruments, Evans drumheads, Rico woodwind reeds and Planet Waves accessories

G.R.A.S.
Sound & Vibration ApS
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Vedbaek, Denmark
Measurement microphones, Intensity probes, Calibrators

Industrial Acoustics Company
www.industrialacoustics.com
Bronx, New York
Research, Engineering and Manufacturing–Products and Services for Noise Control and Acoustically Conditioned Environments

InfoComm International Standards
www.infocomm.org
Fairfax, Virginia
Advancing Audiovisual Communications Globally

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www.ibm.com/us/
Yorktown Heights, New York
Manufacturer of Business Machines

JBL Professional
www.jblpro.com
Northridge, California
Loudspeakers and Transducers of All Types

Knowles Electronics, Inc.
www.knowlesinc.com
Itasca, Illinois
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www.massa.com
Hingham, Massachusetts
Design and Manufacture of Sonar and Ultrasonic Transducers
Computer-Controlled OEM Systems

Meyer Sound Laboratories, Inc.
www.meyersound.com
Berkeley, California
Manufacture Loudspeakers and Acoustical Test Equipment

National Council of Acoustical Consultants
www.ncac.com
Indianapolis, Indiana
An Association of Independent Firms Consulting in Acoustics

Raytheon Company
Integrated Defense Systems
www.raytheon.com
Portsmouth, Rhode Island
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Science Applications International Corporation
Acoustic and Marine Systems Operation
Arlington, Virginia
Underwater Acoustics; Signal Processing; Physical Oceanography; Hydrographic Surveys; Seismology; Undersea and Seismic Systems

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.

Sperian Hearing Protection, LLC
www.howardleight.com
San Diego, California
Howard Leight hearing protection, intelligent protection for military environments, in-ear dosimetry, real-world verification of attenuation, and education supported by the NVLAP-accredited Howard Leight Acoustical Testing Laboratory

Thales Underwater Systems
www.tms-sonar.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/services/arc.html
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 ____________________

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 we should like to indicate our products or services as follows:

____
(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:

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Please send completed applications to: Executive Director, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, (516) 576-2360

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.

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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.

ACOUSTICAL SOCIETY OF AMERICA

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Aprvd by EC________

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)

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Part I Continued ➔

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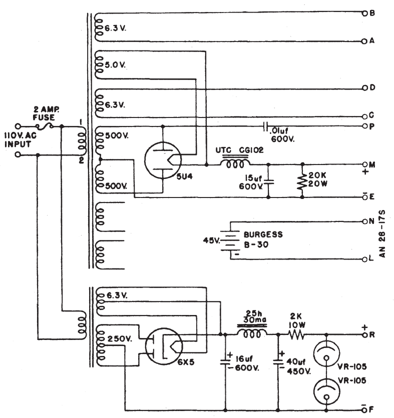

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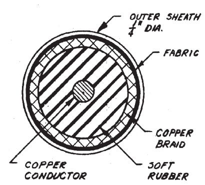

Fig.5. Schematic section showing cable makeup.

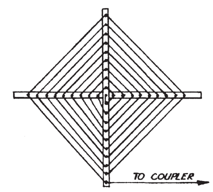

Fig.6. Configuration of the spiral mounting for the cable.

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