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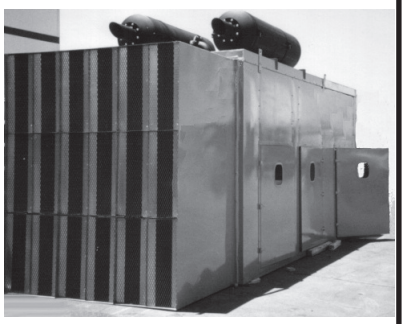

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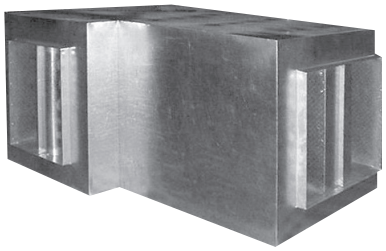

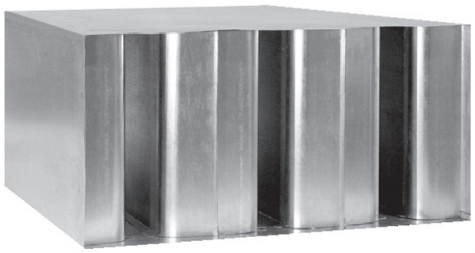

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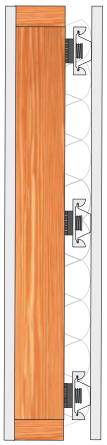

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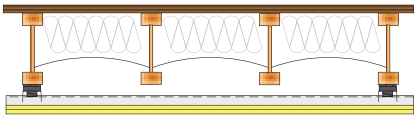

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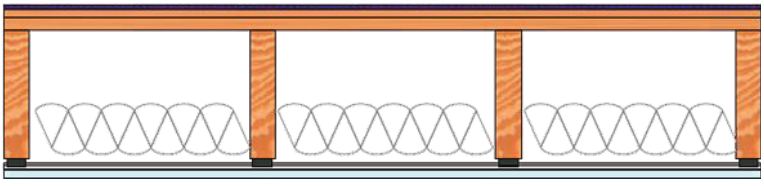

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CONTENTS

page

Technical Program Summary A8
Schedule of Technical Session Starting Times A9
Map of Meeting Rooms at The Galt House A10
Calendar—Technical Program A11
Calendar—Other Events A15
Meeting Information A16
Guidelines for Presentations A22
Dates of Future Meetings A23
50- and 25 Year Anniversaries A24
2018 Annual Giving A25
Technical Sessions (1aAA), Monday Morning 1651
Technical Sessions (1pAO), Monday Afternoon 1670
Technical Session (1eID), Monday Evening 1694
Technical Sessions (2aAAa), Tuesday Morning 1695
Technical Sessions (2pAA), Tuesday Afternoon 1738
Technical Sessions (3aABa), Wednesday Morning 1774
Technical Sessions (3pAA), Wednesday Afternoon 1805
Plenary Session and Awards Ceremony, Wednesday Afternoon 1829
Distinguished Service Citation Encomium 1831
R. Bruce Lindsay Award Encomium 1835
Silver Medal in Engineering Acoustics Encomium 1839
Helmholtz-Rayleigh Interdisciplinary Silver Medal Encomium 1843
Gold Medal Encomium 1847
Technical Session (3eED), Wednesday Evening 1830
Technical Sessions (4aAA), Thursday Morning 1853
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TECHNICAL PROGRAM SUMMARY
177th Meeting of the Acoustical Society of America
13–17 May 2019
***Indicates Special Session**

MONDAY MORNING

*1aAA Integrated Approach to Speech Privacy
*1aAO Future Directions in Acoustical Oceanography
*1aBAa Ultrasound Modeling Workshop
*1aBAb Lung Ultrasound and Tissue Stiffness Method I
*1aNS Acoustics of Healthcare Facilities
*1aPA Battlefield Acoustics I
1aPP Physiology and Modeling
*1aSAa Smart Materials for Acoustics and Vibration I
*1aSAb Vibration Reduction for Extraordinarily Sensitive Applications
*1aSP Reconfigurable Arrays for Adaptive Wave Guiding

MONDAY AFTERNOON

1pAO Topics in Acoustical Oceanography
*1pBA Lung Ultrasound and Tissue Stiffness Method II
*1pMU Transient Phenomena in Wind Instruments
*1pNS Acoustic Vehicle Alerts: Effects on Soundscape, Quality of Life, and Traffic Safety
*1pPAa Battlefield Acoustics II
*1pPAb On His 100th Birthday, Isadore Rudnick Speaks for Himself
*1pPP Applications of Signal Detection Theory in Perception and Physiology
*1pSA Smart Materials for Acoustics and Vibrations II
*1pSC Exploring the Interface Between Linguistic Processing and Talker Recognition
1pUW Target and Radiation by Structures

MONDAY EVENING

*1eID Tutorial Lecture on Computational Methods for Describing Acoustic Propagation in Forests

TUESDAY MORNING

*2aAAa Libraries, Media Centers, and Similar Spaces
*2aAAb Student Design Competition (Poster Session)
*2aAB Bioinspiration and Biomimetics in Acoustics I
*2aBA Cardiovascular Ultrasound: Imaging and Therapy I
*2aID Graduate Programs in Acoustics (Poster Session)
2aMU General Topics in Musical Acoustics
*2aNS Structure-Borne Noise in Buildings and What We Can Do About It
*2aPA Nonlinear Acoustics for Non-Specialists I
*2aPPa Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical Physiological Collaborations
2aPPb Topics in Physiological and Psychoacoustics (Poster Session)
*2aSA Acoustic Metamaterials I
*2aSCa Perception of Speech Directed Toward Infants and Children
*2aSCb Acoustic Phonetic Properties of Speech Directed Toward Infants and Children
2aSPa Beamforming, Detection and Localization
2aSPb Acoustic Detection, Localization and Classification (Poster Session)

TUESDAY AFTERNOON

*2pAA Higher Education Schools of Music
*2pAB Bioinspiration and Biomimetics in Acoustics II
*2pBA Cardiovascular Ultrasound: Imaging and Therapy II
*2pED Acoustics Education Prize Lecture
*2pID Promoting Student Publishing Success
*2pMU Bluegrass Music and Related Instruments
*2pNS Soundscape and its Application Based on the New Standard
*2pPA Nonlinear Acoustics for Non-Specialists II
*2pPPa Auditory Neuroscience Prize Lecture
*2pPPb Cultivating New Growth by Composting Old Ideas: Pruning the Deadwood from the Garden of Psychological and Physiological Acoustics
*2pSA Acoustic Metamaterials II
*2pSC Perception and Production of Speech Directed Toward Infants and Children (Poster Session)
*2pSP Borehole Acoustics Logging for Hydrocarbons Reservoir Characterization
2pUW Reflection and Scattering from Ocean Surface and Bottom

WEDNESDAY MORNING

*3aABa Bioinspiration and Biomimetics in Acoustics III
3aABb Animal Bioacoustics Poster Session
*3aBA Interaction of Light and Ultrasound I
*3aCA Finite Difference Time Domain Method Across Acoustics
*3aED Hands-On Demonstrations
*3aMU Polyphonic Pitch Perception and Analysis I
*3aPA Acoustofluidics I
*3aPP Context Effects in Speech Perception I
3aSAa General Topics in Structural Acoustics and Vibration I
*3aSAb Noise and Vibration in Rotating Machinery
3aSC Developmental & Clinical Populations (Poster Session)
*3aSP Bayesian Inference in Acoustic Signal Processing
3aUW Ocean Acoustics in High Latitudes and General Propagation

WEDNESDAY AFTERNOON

3pAA Acoustical Materials and Testing
3pAB Topics in Animal Bioacoustics
3pBAa Biomedical Acoustics Best Student Paper Competition (Poster Session)
*3pBAb Interaction of Light and Ultrasound II
*3pID Hot Topics in Acoustics
*3pMU Polyphonic Pitch Perception and Analysis II
*3pNS Noise at Sporting Events and Sports Venues
*3pPA Acoustofluidics II
*3pPPa Diversity in Auditory Perception and Speech Communication
3pPPb Context Effects in Speech Perception II (Poster Session)
*3pSA Novel Damping Treatments
3pSC Second Language Speakers and Listeners (Poster Session)

WEDNESDAY EVENING

*3eED Listen Up and Get Involved

THURSDAY MORNING

*4aAA Methods and Techniques Used for Simulation of Room Acoustics
4aAB Marine Mammal Bioacoustics
*4aBAa Inverse Problems in Biomedical Ultrasound I
4aBAb General Topics in Biomedical Acoustics I
4aEA General Topics in Engineering Acoustics: Sensors and Sources
*4aNS Increasing Noise Awareness in Society
*4aPA Infrasound I
4aPP Spatial Hearing, Complex Acoustic Scenes, and Clinical Devices (Poster Session)
4aSA General Topics in Structural Acoustics and Vibration II
4aSP Emerging Techniques for Acoustic Signal Processing
*4aUW Uncertainty in Propagation Prediction

THURSDAY AFTERNOON

*4pAA Room Acoustics Modeling and Auralization
*4pBAa Inverse Problems in Biomedical Ultrasound II
4pBAb General Topics in Biomedical Acoustics II
4pEA General Topics in Engineering Acoustics: Characterization and Measurement
*4pNS Advances and Applications in Sound Quality Metrics
*4pPAa Infrasound II
4pPAb General Topics In Physical Acoustics I
*4pPP Perceptual Consequences of Hearing Loss Across the Lifespan: From Children to Adults (Physiology Meets Perception)
4pSC Perception (Poster Session)

FRIDAY MORNING

*5aAA Restaurant Acoustics
*5aAB Understanding Animal Song
5aBA General Topics in Biomedical Acoustics III
5aPA General Topics in Physical Acoustics II
5aSC Production (Poster Session)
5aUW Underwater Signal Processing and Applications

SCHEDULE OF STARTING TIMES FOR TECHNICAL SESSIONS AND TECHNICAL COMMITTEE/TECHNICAL SPECIALITY GROUP (TC/TSG) MEETINGS

	Monday AM	Monday PM	Monday Eve	Tuesday AM	Tuesday PM	Tuesday Eve	Wednesday AM	Wednesday PM	Wednesday Eve	Thursday AM	Thursday PM	Thursday Eve	Friday AM
Beckham	1aSP 9:00			2aSPa 8.00	2pSP 1:30	TCSP 4:30	3aSP 8:30			4aSP 8:30			
Breathitt		1pMU 2:00		2aMU 9:00	2pID 1:00	TCMU 7:30	3aMU 8:00	3pMU 1:00		4aBAb 8:00			
Carroll Ford	1aPP 10:10	1pPP 1:05		2aPPa 8:00	2pPPa 1:00 2pPPb 2:10	TCPP 7:30	3aPP 8:00	3pPPa 1:30			4pPP 1:15	TCSC 7:30	
Clements			1eID 7:00	2aAB 8:30	2pAB 1:00	TCAB 7:30	3aABa 8:00	3pAB 1:30		4aAB 8:30		TCCA 4:30	5aAB 8:00
Coe	1aSAb 9:10				2pED 3:25		3aED 8:30		3eED 5:00				
Combs Chandler		1pSC 1:00		2aSCa 8:00 2aSCb 10:15	2pMU 3:00			3pID 1:00					
Exhibit Hall				2aAAb 8:00 2aID 8:00				3pBAa 1:00					
French	1aAA 9:30			2aAAa 8:00	2pAA 1:30	TCAA 7:30	3aCA 9:00	3pAA 1:00		4aAA 8:15	4pAA 1:10		5aAA 8:00
Grand C				2aPPb 9:00 2aSPb 10.45	2pSC 1:30		3aABb 10:00 3aSC 9:00	3pPPb 1:30 3pSC 1:30		4aPP 8:00	4pSC 1:30		5aSC 8:00
Jones	1aPA 8:40	1pPAa 1:00 1pPAb 3:15		2aPA 7:55	2pPA 1:30	TCPA 7:30	3aPA 8:30	3pPA 1:00		4aPA 8:05	4pPAa 1:00		5aPA 8:15
McCreary		1pUW 1:30			2pUW 1:30	TCAO 7:30	3aUW 9:00			4aUW 8:20	4pPAb 3:15	TCUW 7:30	5aUW 9:00
Nunn	1aBAa 8:30 1aBAb 11:00	1pBA 1:20		2aBA 8:30	2pBA 1:30	TCEA 4:30	3aBA 9:00	3pBAb 1:20	TCBA 7:30	4aBAa 7:55	4pBAa 1:00 4pBAb 1:55		5aBA 8:00
Segell	1aNS 9:00	1pNS 1:30		2aNS 8:00	2pNS 1:30		3aSAb 10:45	3pNS 1:20		4aNS 8:50	4pNS 1:05	TCNS 7:30	
Stopher	1aSAa 8:00	1pSA 1:30		2aSA 8:00	2pSA 1:15	TCSA 7:30	3aSAa 8:00	3pSA 1:00		4aSA 8:00			
Willis	1aAO 8:30	1pAO 1:00								4aEA 8:00	4pEA 1:30		

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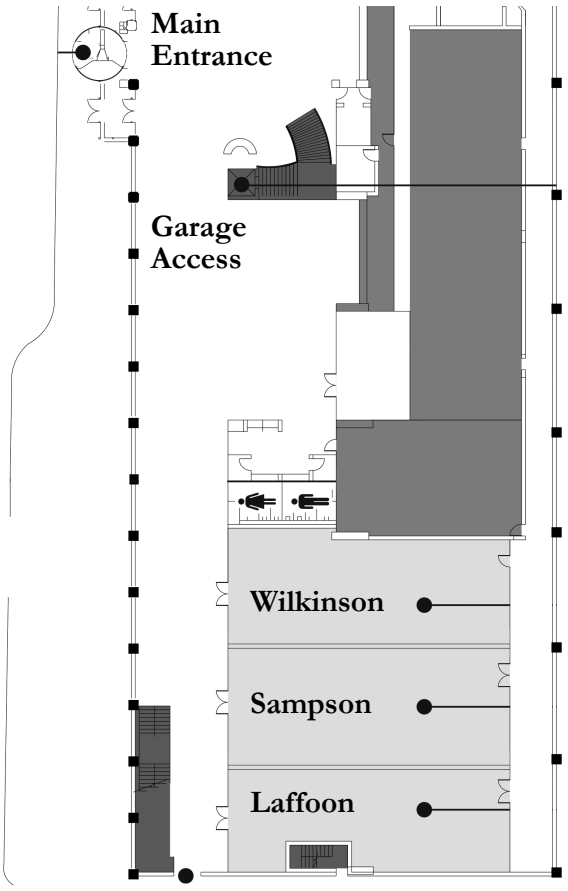

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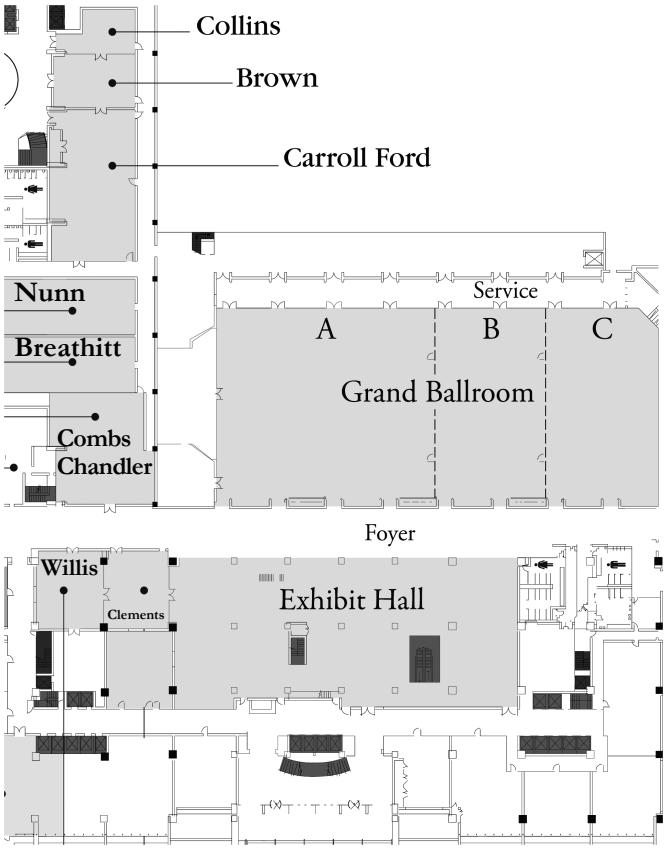

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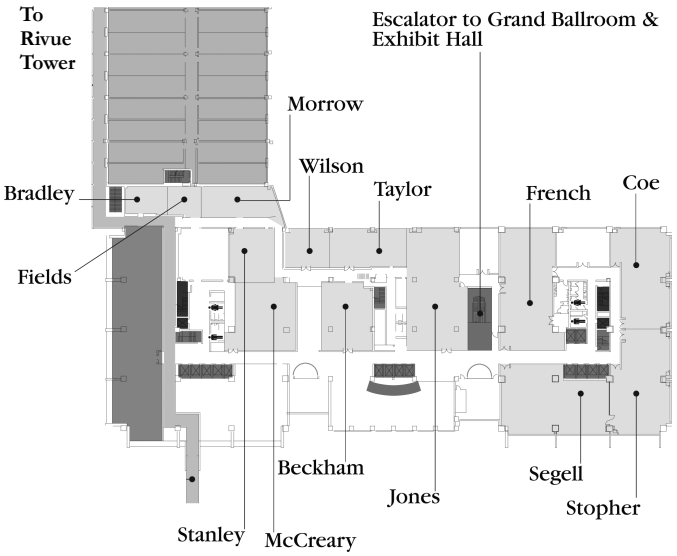

TECHNICAL PROGRAM CALENDAR
177th Meeting of the Acoustical Society of America
13–17 May 2019

MONDAY MORNING

9:30 1aAA **Architectural Acoustics, Noise, Signal Processing in Acoustics, and Speech Communication:** Integrated Approach to Speech Privacy. French

8:30 1aAO **Acoustical Oceanography:** Future Directions in Acoustical Oceanography. Willis

8:30 1aBAa **Biomedical Acoustics:** Ultrasound Modeling Workshop. Nunn

11:00 1aBAb **Biomedical Acoustics and Signal Processing in Acoustics:** Lung Ultrasound and Tissue Stiffness Method I. Nunn

9:00 1aNS **Noise, Architectural Acoustics, and ASA Committee on Standards:** Acoustics of Healthcare Facilities. Segell

8:40 1aPA **Physical Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, Noise, Psychological and Physiological Acoustics, and Speech Communication:** Battlefield Acoustics I. Jones

10:10 1aPP **Psychological and Physiological Acoustics:** Physiology and Modeling. Carroll Ford

8:00 1aSAa **Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Architectural Acoustics:** Smart Materials for Acoustics and Vibration I. Stopher

9:10 1aSAb **Structural Acoustics and Vibration and Architectural Acoustics:** Vibration Reduction for Extraordinarily Sensitive Applications. Coe

9:00 1aSP **Signal Processing in Acoustics, Engineering Acoustics, Physical Acoustics, Underwater Acoustics, and Structural Acoustics and Vibration:** Reconfigurable Arrays for Adaptive Wave Guiding. Beckham

MONDAY AFTERNOON

1:00 1pAO **Acoustical Oceanography:** Topics in Acoustical Oceanography. Willis

1:20 1pBA **Biomedical Acoustics and Signal Processing in Acoustics:** Lung Ultrasound and Tissue Stiffness Method II. Nunn

2:00 1pMU **Musical Acoustics and Signal Processing in Acoustics:** Transient Phenomena in Wind Instruments. Breathitt

1:30 1pNS **Noise and Psychological and Physiological Acoustics:** Acoustic Vehicle Alerts: Effects on Soundscape, Quality of Life, and Traffic Safety. Segell

1:00 1pPAa **Physical Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, Noise, Psychological and Physiological Acoustics, and Speech Communication:** Battlefield Acoustics II. Jones

3:15 1pPAb **Physical Acoustics, Archives and History, and Education in Acoustics:** On His 100th Birthday, Isadore Rudnick Speaks for Himself. Jones

1:05 1pPP **Psychological and Physiological Acoustics, Speech Communication, and Signal Processing in Acoustics:** Applications of Signal Detection Theory in Perception and Physiology. Carroll Ford

1:30 1pSA **Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Architecture Acoustics:** Smart Materials for Acoustics and Vibrations II. Stopher

1:00 1pSC **Speech Communication and Signal Processing in Acoustics:** Exploring the Interface Between Linguistic Processing and Talker Recognition. Combs Chandler

1:30 1pUW **Underwater Acoustics:** Target and Radiation by Structures. McCreary

MONDAY EVENING

7:00 1eID **Interdisciplinary:** Tutorial Lecture on Computational Methods for Describing Acoustic Propagation in Forests. Clements

TUESDAY MORNING

8:00 2aAAa **Architectural Acoustics and Noise:** Libraries, Media Centers, and Similar Spaces. French

8:00 2aAAb **Architectural Acoustics:** Student Design Competition (Poster Session). Exhibit Hall

8:30 2aAB **Animal Bioacoustics and Signal Processing in Acoustics:** Bioinspiration and Biomimetics in Acoustics I. Clements

8:30 2aBA **Biomedical Acoustics and Signal Processing in Acoustics:** Cardiovascular Ultrasound: Imaging and Therapy I. Nunn

8:00 2aID **Interdisciplinary and Student Council:** Graduate Programs in Acoustics (Poster Session). Exhibit Hall

9:00 2aMU **Musical Acoustics:** General Topics in Musical Acoustics. Breathitt

8:00 2aNS **Noise, Architectural Acoustics, Structural Acoustics and Vibration, and ASA Committee on Standards:** Structure-Borne Noise in Buildings and What We Can Do About It. Segell

7:55 2aPA **Physical Acoustics and Noise:** Nonlinear Acoustics for Non-Specialists I. Jones

8:00 2aPPa **Psychological and Physiological Acoustics and Education in Acoustics:** Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical Physiological Collaborations. Carroll Ford

9:00 2aPPb **Psychological and Physiological Acoustics:** Topics in Physiological and Psychoacoustics (Poster Session). Grand Ballroom C

8:00 2aSA **Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, Noise, and Architectural Acoustics:** Acoustic Metamaterials I. Stopher

8:00 2aSCa **Speech Communication:** Perception of Speech Directed Toward Infants and Children. Combs Chandler

10:15 2aSCb **Speech Communication and Psychological and Physiological Acoustics:** Acoustic Phonetic Properties of Speech Directed Toward Infants and Children. Combs Chandler

8:00 2aSPa **Signal Processing in Acoustics:** Beamforming, Detection and Localization. Beckham

10:45 2aSPb **Signal Processing in Acoustics:** Acoustic Detection, Localization and Classification (Poster Session). Grand Ballroom C

TUESDAY AFTERNOON

1:30 2pAA **Architectural Acoustics, Musical Acoustics, and Education in Acoustics:** Higher Education Schools of Music. French

1:00 2pAB **Animal Bioacoustics and Signal Processing in Acoustics:** Bioinspiration and Biomimetics in Acoustics II. Clements

1:30 2pBA **Biomedical Acoustics and Signal Processing in Acoustics:** Cardiovascular Ultrasound: Imaging and Therapy II. Nunn

3:25 2pED **Education in Acoustics:** Acoustics Education Prize Lecture. Coe

1:00 2pID **Interdisciplinary and Education in Acoustics:** Promoting Student Publishing Success. Breathitt

3:00 2pMU **Musical Acoustics:** Bluegrass Music and Related Instruments. Combs Chandler

1:30 2pNS **Noise, Architectural Acoustics, ASA Committee on Standards, and Animal Bioacoustics:** Soundscape and its Application Based on the New Standard. Segell

1:30 2pPA **Physical Acoustics and Noise:** Nonlinear Acoustics for Non-Specialists II. Jones

1:00 2pPPa **Psychological and Physiological Acoustics:** Auditory Neuroscience Prize Lecture. Carroll Ford

2:10 2pPPb **Psychological and Physiological Acoustics and Education in Acoustics:** Cultivating New Growth by Composting Old Ideas: Pruning the Deadwood from the Garden of Psychological and Physiological Acoustics. Carroll Ford

1:15 2pSA **Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, Noise, and Architectural Acoustics:** Acoustic Metamaterials II. Stopher

1:30 2pSC **Speech Communication and Psychological and Physiological Acoustics:** Perception and Production of Speech Directed Toward Infants and Children (Poster Session). Grand Ballroom C

1:30 2pSP **Signal Processing in Acoustics, Acoustical Oceanography, Physical Acoustics, Underwater Acoustics, and Structural Acoustics and Vibration:** Borehole Acoustics Logging for Hydrocarbons Reservoir Characterization. Beckham

1:30 2pUW **Underwater Acoustics:** Reflection and Scattering from Ocean Surface and Bottom. McCreary

WEDNESDAY MORNING

8:00 3aABa **Animal Bioacoustics and Signal Processing in Acoustics:** Bioinspiration and Biomimetics in Acoustics III. Clements

10:00 3aABb **Animal Bioacoustics:** Animal Bioacoustics Poster Session. Grand Ballroom C

9:00 3aBA **Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics:** Interaction of Light and Ultrasound I. Nunn

9:00 3aCA **Computational Acoustics, Biomedical Acoustics, Physical Acoustics, Underwater Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration:** Finite Difference Time Domain Method Across Acoustics. French

8:30 3aED **Education in Acoustics:** Hands-On Demonstrations. Coe

8:00 3aMU **Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics:** Polyphonic Pitch Perception and Analysis I. Breathitt

8:30 3aPA **Physical Acoustics, Engineering Acoustics, and Biomedical Acoustics:** Acoustofluidics I. Jones

8:00 3aPP **Psychological and Physiological Acoustics and Speech Communication:** Context Effects in Speech Perception I. Carroll Ford

8:00 3aSAa **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration I. Stopher

10:45 3aSAb **Structural Acoustics and Vibration, Engineering Acoustics, and Noise:** Noise and Vibration in Rotating Machinery. Segell

9:00 3aSC **Speech Communication:** Developmental and Clinical Populations (Poster Session). Grand Ballroom C

8:30 3aSP **Signal Processing in Acoustics, Underwater Acoustics, Acoustical Oceanography, and Noise:** Bayesian Inference in Acoustic Signal Processing. Beckham

9:00 3aUW **Underwater Acoustics and Acoustical Oceanography:** Ocean Acoustics in High Latitudes and General Propagation. McCreary

WEDNESDAY AFTERNOON

1:00 3pAA **Architectural Acoustics:** Acoustical Materials and Testing. French

1:30 3pAB **Animal Bioacoustics:** Topics in Animal Bioacoustics. Clements

1:00 3pBAa **Biomedical Acoustics:** Biomedical Acoustics Best Student Paper Competition (Poster Session). Exhibit Hall

1:20 3pBAb **Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics:** Interaction of Light and Ultrasound II. Nunn

1:00 3pID **Interdisciplinary:** Hot Topics in Acoustics. Combs Chandler

1:00 3pMU **Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics:** Polyphonic Pitch Perception and Analysis II. Breathitt

1:20 3pNS **Noise, Architectural Acoustics, and Structural Acoustics and Vibration:** Noise at Sporting Events and Sports Venues. Segell

1:00 3pPA **Physical Acoustics, Engineering Acoustics, and Biomedical Acoustics:** Acoustofluidics II. Jones

1:30 3pPPa **Psychological and Physiological Acoustics, Speech Communication, and Education in Acoustics:** Diversity in Auditory Perception and Speech Communication. Carroll Ford

1:30 3pPPb **Psychological and Physiological Acoustics and Speech Communication:** Context Effects in Speech Perception II (Poster Session). Grand Ballroom C

1:00 3pSA **Structural Acoustics and Vibration, Engineering Acoustics, Noise, and Architectural Acoustics:** Novel Damping Treatments. Stopher

1:30 3pSC **Speech Communication:** Second Language Speakers and Listeners (Poster Session). Grand Ballroom C

WEDNESDAY EVENING

5:00 3eED **Education in Acoustics and Women in Acoustics:** Listen Up and Get Involved. Coe

THURSDAY MORNING

8:15 4aAA **Architectural Acoustics, Signal Processing in Acoustics, and Noise:** Methods and Techniques Used for Simulation of Room Acoustics. French

8:30 4aAB **Animal Bioacoustics:** Marine Mammal Bioacoustics. Clements

7:55 4aBAa **Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics:** Inverse Problems in Biomedical Ultrasound I. Nunn

8:00 4aBAb **Biomedical Acoustics:** General Topics in Biomedical Acoustics I. Breathitt

8:00 4aEA **Engineering Acoustics:** General Topics in Engineering Acoustics: Sensors and Sources. Willis

8:50 4aNS **Noise and Education in Acoustics:** Increasing Noise Awareness in Society. Segell

8:05 4aPA **Physical Acoustics and Signal Processing in Acoustics:** Infrasound I. Jones

8:00 4aPP **Psychological and Physiological Acoustics and Speech Communication:** Spatial Hearing, Complex Acoustic Scenes, and Clinical Devices (Poster Session). Grand Ballroom C

8:00 4aSA **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration II. Stopher

8:30 4aSP **Signal Processing in Acoustics:** Emerging Techniques for Acoustic Signal Processing. Beckham

8:20 4aUW **Underwater Acoustics, Physical Acoustics, Structural Acoustics and Vibration, Acoustical Oceanography, Signal Processing in Acoustics, and Computational Acoustics:** Uncertainty in Propagation Prediction. McCreary

THURSDAY AFTERNOON

1:10 4pAA **Architectural Acoustics and Signal Processing in Acoustics:** Room Acoustics Modeling and Auralization. French

1:00 4pBAa **Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics:** Inverse Problems in Biomedical Ultrasound II. Nunn

1:55 4pBAb **Biomedical Acoustics:** General Topics in Biomedical Acoustics II. Nunn

1:30 4pEA **Engineering Acoustics:** General Topics in Engineering Acoustics: Characterization and Measurement. Willis

1:05 4pNS **Noise, Signal Processing in Acoustics, and Psychological and Physiological Acoustics:** Advances and Applications in Sound Quality Metrics. Segell

1:00 4pPAa **Physical Acoustics and Signal Processing in Acoustics:** Infrasound II. Jones

3:15 4pPAb **Physical Acoustics:** General Topics In Physical Acoustics I. McCreary

1:15 4pPP **Psychological and Physiological Acoustics and Speech Communication:** Perceptual Consequences of Hearing Loss Across the Lifespan: From Children to Adults (Physiology Meets Perception). Carroll Ford

1:30 4pSC **Speech Communication:** Perception (Poster Session). Grand Ballroom C

FRIDAY MORNING

8:00 5aAA **Architectural Acoustics, Noise, and ASA Committee on Standards:** Restaurant Acoustics. French

8:00 5aAB **Animal Bioacoustics, Signal Processing in Acoustics, Acoustical Oceanography, Speech Communication, and Underwater Acoustics:** Understanding Animal Song. Clements

8:00 5aBA **Biomedical Acoustics:** General Topics in Biomedical Acoustics III. Nunn

8:15 5aPA **Physical Acoustics:** General Topics in Physical Acoustics II. Jones

8:00 5aSC **Speech Communication:** Production (Poster Session). Grand Ballroom C

9:00 5aUW **Underwater Acoustics:** Underwater Signal Processing and Applications. McCreary

SCHEDULE OF COMMITTEEE MEETINGS AND OTHER EVENTS

All rooms are located in the Suite Tower

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Mon, 13 May, 7:30 a.m.	Executive Council	Brown
Mon, 13 May, 3:30 p.m.	Technical Council	Brown
Mon, 13 May, 6:30 p.m.	Technical Council Dinner	Sampson
Tue, 14 May, 7:00 a.m.	ASA Books	Fields
Tue, 14 May, 7:00 a.m.	POMA Editorial Board	Wilson
Tue, 14 May, 7:30 a.m.	Panel on Public Policy	Taylor
Tue, 14 May, 11:45 a.m.	Editorial Board	Sampson
Tue, 14 May, 12:00 noon	Student Council	Fields
Tue, 14 May, 12:30 p.m.	Prizes & Special Fellowships	Bradley
Tue, 14 May, 1:30 p.m.	Meetings	Morrow
Tue, 14 May, 4:00 p.m.	International Liaison	Bradley
Tue, 14 May, 4:00 p.m.	Newman Fund Advisory	Fields
Tue, 14 May, 4:30 p.m.	Education in Acoustics	McCreary
Tue, 14 May, 5:00 p.m.	Women in Acoustics	Morrow
Wed, 15 May, 7:00 a.m.	Archives & History	Fields
Wed, 15 May, 7:00 a.m.	College of Fellows	Morrow
Wed, 15 May, 7:00 a.m.	International Research & Education	Bradley
Wed, 15 May, 7:00 a.m.	Publication Policy	Taylor
Wed, 15 May, 7:00 a.m.	Regional and Student Chapters	Wilkinson
Wed, 15 May, 7:30 a.m.	Finance	Brown
Wed, 15 May, 9:30 a.m.	Foundation Board	Fields
Wed, 15 May, 11:00 a.m.	Medals and Awards	Lafoon
Wed, 15 May, 11:30 a.m.	Public Relations	Morrow
Wed, 15 May, 12:00 noon	Audit	Bradley
Wed, 15 May, 12:00 noon	Membership	Fields
Wed, 15 May, 5:00 p.m.	TCAA Speech Privacy	Taylor
Thu, 16 May, 7:00 a.m.	Investments	Wilson
Thu, 16 May, 7:30 a.m.	Tutorials, Short Courses, Hot Topics	Fields
Thu, 16 May, 9:00 a.m	Financial Affairs Admin Council	Wilson
Thu, 16 May, 2:00 p.m.	Strategic Plan Champions	Wilkinson
Thu, 16 May, 4:30 p.m.	Member Engagement and Diversity	Lafoon
Thu, 16 May, 4:30 p.m.	Outreach	Wilson
Thu, 16 May, 4:30 p.m.	Publications and Standards	Fields
Fri, 17 May, 7:00 a.m.	Technical Council	Brown
Fri, 17 May, 11:00 a.m.	Executive Council	Brown

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 14 May, 4:30 p.m.	Engineering Acoustics	Nunn
Tue, 14 May, 4:30 p.m.	Signal Processing in Acoustics	Beckham
Tue, 14 May, 7:30 p.m.	Acoustical Oceanography	McCreary
Tue, 14 May, 7:30 p.m.	Animal Bioacoustics	Clements
Tue, 14 May, 7:30 p.m.	Architectural Acoustics	French
Tue, 14 May, 7:30 p.m.	Musical Acoustics	Breathitt
Tue, 14 May, 7:30 p.m.	Physical Acoustics	Jones
Tue, 14 May, 7:30 p.m.	Psychological and Physiological Acoustics	Carroll Ford
Tue, 14 May, 7:30 p.m.	Structural Acoustics and Vibration	Stopher
Wed, 15 May, 7:30 p.m.	Biomedical Acoustics	Nunn
Thu, 16 May, 4:30 p.m.	Computational Acoustics	Clements
Thu, 16 May, 7:30 p.m.	Noise	Segell
Thu, 16 May, 7:30 p.m.	Speech Communication	Carroll Ford
Thu, 16 May, 7:30 p.m.	Underwater Acoustics	McCreary

STANDARDS COMMITTEES AND WORKING GROUPS

Mon, 13 May, 10:00 a.m.	WG58-Small Unmanned Systems	Bradley
Mon, 13 May, 4:30 p.m.	WG18-Room Noise	Morrow
Mon, 13 May, 5:00 p.m.	S2-Mechanical Vibration and Shock	Wilson
Mon, 13 May, 7:00 p.m.	ASACOS Steering	Wilson
Tue, 14 May, 7:30 a.m.	ASACOS	Morrow
Tue, 14 May, 9:15 a.m.	Standards Plenary Group	Morrow
Tue, 14 May, 11:00 a.m.	S1-Acoustics	Morrow
Tue, 14 May, 2:00 p.m.	S3 Bioacoustics	Taylor
Tue, 14 May, 3:30 p.m.	S3/SC1-Animal Bioacoustics	Taylor
Tue, 14 May, 5:00 p.m.	S12-Noise	Taylor
Tue, 14 May, 5:00 p.m.	WG44-Speech Privacy in Healthcare	Wilson

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Sun, 12 May 1:00 p.m. - 5:00 p.m. Mon, 13 May 8:00 a.m. - 12:30 p.m.	Short Course	Clements
Mon-Thu, 13-16 May 7:30 a.m. - 5:00 p.m. Fri, 17 May 7:30 a.m. - 12:00 noon	Registration	Exhibit Hall (2nd Fl)
Mon-Thu, 13-16 May, 7:00 a.m. - 5:00 p.m. Fri, 17 May, 7:00 a.m. - 12:00 noon	E-mail	Grand Foyer
Mon-Thu, 13-16 May, 7:00 a.m. - 5:00 p.m. Fri, 17 May, 7:00 a.m. - 12:00 noon	Internet Zone	Grand Foyer
Sun, 12 May, 3:00 p.m. - 5:00 p.m. Mon-Thu, 13-16 May, 7:00 a.m. - 5:00 p.m. Fri, 17 May 7:00 a.m. - 12:00 noon	A/V Preview	Stanley
Mon-Fri, 13-17 May 8:00 a.m. - 10:00 a.m.	Accompanying Persons	Terrace 16th Fl.
Mon-Fri, 13-17 May 9:00 a.m. - 5:00 p.m.	Gallery of Acoustics	Grand Foyer
Mon-Fri, 13-17 May, 9:45 a.m. - 10:45 a.m.	Coffee Break	Grand Foyer
Mon, 13 May 3:30 p.m. - 5:00 p.m.	Early Career Publishing Workshop	Sampson
Mon, 13 May 5:00 p.m. - 5:30 p.m.	New Student Orientation	Willis
Mon, 13 May 5:30 p.m. - 7:00 p.m.	Student Meet and Greet	Wilkinson
Tue, 14 May 1:30 p.m. - 2:30 p.m.	Women in Acoustics Roundtable	Wilkinson
Tue, 14 May 6:00 p.m. - 7:30 p.m.	Social Hour	Grand A/B
Wed, 15 May, 11:45 a.m. - 1:45 p.m.	Women in Acoustics Luncheon	Sampson
Wed, 15 May, 3:30 p.m. - 5:00 p.m.	Plenary Session/Awards Ceremony	Grand B
Wed, 15 May, 6:00 p.m. - 8:00 p.m.	Student Reception	Waterford 25th Fl.
Wed, 15 May, 8:00 p.m. - 12:00 midnight	ASA Jam	TBD
Thu, 16 May, 12:00 noon - 2:00 p.m.	Society Luncheon and Lecture	Sampson
Thu, 16 May, 6:00 p.m. - 7:30 p.m.	Social Hour	Grand A/B

177th Meeting of the Acoustical Society of America

The 177th meeting of the Acoustical Society of America will be held Monday through Friday, 13–17 May 2019 at The Galt House, Louisville, Kentucky, USA.

SECTION HEADINGS

1. HOTEL INFORMATION
2. TRANSPORTATION AND TRAVEL
3. MESSAGES FOR ATTENDEES
4. REGISTRATION
5. ACCESSIBILITY
6. TECHNICAL SESSIONS
7. TECHNICAL SESSION DESIGNATIONS
8. HOT TOPICS SESSION
9. ULTRASOUND MODELING WORKSHOP
10. EARLY CAREER PUBLISHING WORKSHOP
11. WOMEN IN ACOUSTICS ROUND-TABLE DISCUSSION
12. ROSSING PRIZE IN ACOUSTICS EDUCATION AND THE EDUCATION IN ACOUSTICS PRIZE LECTURE
13. WILILAM AND CHRISTINE PRIZE IN AUDITORY NEUROSCIENCE AND THE AUDITORY NEUROSCIENCE PRIZE LECTURE
14. TUTORIAL LECTURE
15. SHORT COURSE
16. GALLERY OF ACOUSTICS
17. TECHNICAL COMMITTEE OPEN MEETINGS
18. PLENARY SESSION AND AWARDS CEREMONY
19. ANSI STANDARDS COMMITTEES
20. COFFEE BREAKS
21. A/V PREVIEW ROOM
22. PROCEEDINGS OF MEETINGS ON ACOUSTICS
23. E-MAIL AND INTERNET ZONE
24. SOCIALS
25. SOCIETY LUNCHEON AND LECTURE
26. STUDENT EVENTS: NEW STUDENT ORIENTATION, MEET AND GREET, FELLOWSHIP AND GRANT PANEL, STUDENT RECEPTION
27. WOMEN IN ACOUSTICS LUNCHEON
28. JAM SESSION
29. ACCOMPANYING PERSONS PROGRAM
30. WEATHER
31. TECHNICAL PROGRAM ORGANIZING COMMITTEE
32. MEETING ORGANIZING COMMITTEE
33. PHOTOGRAPHING AND RECORDING
34. ABSTRACT ERRATA
35. GUIDELINES FOR ORAL PRESENTATIONS
36. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS
37. GUIDELINES FOR USE OF COMPUTER PROJECTION
38. DATES OF FUTURE ASA MEETINGS

1. HOTEL INFORMATION

The Galt House is the headquarters hotel where meeting events will be held.

The cut-off date for reserving rooms at special rates has passed. Please contact The Galt House (800-843-4258 (502-589-5200, 40 N. Fourth Street, Louisville, KY 40202) for information about room availability.

2. TRANSPORTATION AND TRAVEL

Air Transportation

The Louisville International Airport (SDF) is 10-minutes from downtown and serves the Kentucky and Southern Indiana region. The airport offers nonstop service to 31 destinations and connections to cities worldwide. While several airlines service SDF, Southwest Airlines is the predominant carrier. For a complete listing of airlines that service SDF visit www.flylouisville.com.

Ground Transportation

Taxi: Cabs are available at the traffic island on the left of the taxi stand at the airport. (Ask about Share-a-Ride at the taxi stand.):
Taxi7 – (502) 777-7777
Yellow Cab – (502) 636-5511
Downtown Louisville is approximately 7 miles from the airport, and taxi fares are approximately $20 for up to 4 people to downtown.

Uber, Lyft, app-based ride service: Lyft and Uber are the only authorized ridesharing services available to transport passengers from the airport. The Lyft and Uber pick-up curb is located on the lower level, east side of the terminal on the inner curb.

Airport Shuttle: Airport transportation can be arranged by appointment through Xtreme Transportation, The Galt House transportation partner. They can be contacted at 502-561-4022 or via Email: galthouseshuttle@xtlimo.com

Car Rental: Rental counters are located on the lower level near baggage claim. Transportation is provided outside on the lower level. The following rental car companies service the Louisville International Airport- Alamo, Avis, Budget, Dollar, Enterprise, Hertz, National, Payless and Thrifty.

Louisville is centrally located at the intersection of Interstate 65 from the north and south, Interstate 64 from the east and west. Cincinnati, Indianapolis, and Nashville are within 3 hours by car. Chicago, Detroit, Pittsburgh, St. Louis, and Memphis are within 6 hours.

Parking: The Galt House valet parking cost is $28/day, self-parking is $20/day. For more information on hotel parking, visit The Galt House webpage.

For additional downtown parking, please visit louisvilledowntown.org/parking-map/ for locations, availability and pricing

3. MESSAGES FOR ATTENDEES

A message board will be located in the Exhibit Hall near the ASA registration desk. Check the board during the week as messages may be posted by attendees who do not have cell phone numbers of other attendees.

4. 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, 13 May, at 7:30 a.m. in the Exhibit Hall (see floor plan on page A10).

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 $650 for members of the Acoustical Society of America; $800 for non-members, $200 for Emeritus members (Emeritus status approved by ASA before the meeting), $375 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $150 for ASA Student members, $250 for students who are not members of ASA, $25 for undergraduate students, and $200 for accompanying persons.

One-day registration is available at $375 for members and $450 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 $800 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 2019 dues.

Invited speakers who are members of the Acoustical Society of America are expected to pay the Member full-week or one-day registration fees. 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 $450 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. If you do not have student id, you will be required to pay the regular registration fee.

5. ACCESSIBLITY

If you have special accessibility requirements, please indicate this below by informing ASA (1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; asa@acousticalsociety.org) at a minimum of thirty days in advance of the meeting. Please provide a cell phone number, email address, and detailed information including the nature of the special accessibility so that we may contact you directly.

6. TECHNICAL SESSIONS

The technical program includes 102 sessions with 998 abstracts scheduled for presentation during the meeting.

A floor plan of The Galt House appears on page A10. 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.

7. TECHNICAL SESSION DESIGNATIONS

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

1-Monday, 13 May
2-Tuesday, 14 May
3-Wednesday, 15 May
4-Thursday, 16 May
5-Friday, 17 May

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

AA	Architectural Acoustics
AB	Animal Bioacoustics
AO	Acoustical Oceanography
BA	Biomedical Acoustics
CA	Computational 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 begins earlier in the same morning.

8. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 15 May, at 1:00 p.m. in the Combs Chandler Room. Papers will be presented on current topics in the fields of Physical Acoustics, Biomedical Acoustics, and Computational Acoustics.

9. ULTRASOUND MODELING WORKSHOP

A 2-hour hands-on workshop using FOCUS, the ‘Fast Object-oriented C++ Ultrasound Simulator’ will be offered

at the Louisville ASA meeting on Monday 13 May in Session 1aBAa at 8.30 a.m. in the Nunn room. This workshop is sponsored by the Biomedical Acoustics Technical Committee and will be available to all who are interested. There is no fee to participate, however, at-meeting registration is subject to availability of space in the workshop.

10. EARLY CAREER PUBLISHING WORKSHOP3

The Workshop will be held on Monday, 13 May, from 3:30 p.m. to 5:00 p.m. in the Sampson Room.

In this workshop, participants will have the opportunity to meet and talk with the current Editor-in-Chief and Associate Editors from the Journal of the Acoustical Society of America (JASA). The Associate Editors will each lead small group discussions about the submission and review process and will field participant questions.

The Early Career Publishing Workshop is intended for early career acousticians from any subfield of acoustics, who received their last degree within the past ten years. The event is not intended for students or those in the processing of receiving a degree.

11. WOMEN IN ACOUSTICS ROUND-TABLE DISCUSSIONS

The Women in Acoustics Committee is hosting a facilitated round-table discussion session from 1:30 p.m. to 2:30 p.m. on Tuesday, 14 May, in the Wilkinson Room.

Discussion topics will include navigating careers in academia, government, and industry; mentoring at all levels; work-life balance; and navigating power differentials. Topic leaders will facilitate the informal discussions, and the attendees may choose which topic they would like to discuss. There will be an opportunity for attendees to switch at 2:00 p.m. to discuss a new topic. While the discussions in this session will focus on women's experiences related to these topics, anyone interested in participating in these discussions is welcome to attend.

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

The Rossing Prize in Acoustics Education will be presented to Stanley Chin-Bing at the Plenary Session on Wednesday, 15 May. Dr. Chin-Bing will present the Acoustics Education Prize Lecture titled "The University of New Orleans ocean acoustics program at the Stennis Space Center, Mississippi" on Tuesday, 14 May, at 3:25 p.m. in Session 2pED in the Coe Room.

13. WILLIAM AND CHRISTINE HARTMANN PRIZE IN AUDITORY NEUROSCIENCE AND AUDITORY NEUROSCIENCE PRIZE LECTURE

The 2019 William and Christine Prize in Auditory Neuroscience will be presented to Glenis Long, City University of New York Graduate Center, at the Plenary Session on Wednesday, 15 May. Dr. Long will present the Auditory Neuroscience Prize Lecture titled "Differences and similarities of peripheral auditory systems" on Tuesday, 14 May, at 1:00 p.m. in Session 2pPPa in the Carroll Ford Room.

14. TUTORIAL LECTURE ON COMPUTATIONAL METHODS FOR DESCRIBING ACOUSTIC PROPAGATION IN FORESTS

A tutorial presentation titled "Computational Methods for Describing Acoustic Propagation in Forests" will be presented by Michelle Swearingen, U. S. Army Engineer Research and Development Center/Construction Engineering Research Laboratory, on Monday, 13 May at 7:00 p.m. in the Clements Room.

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

The registration fee is USD $25 (USD $12 for students with current student IDs).

15. SHORT COURSE ON ELECTRONIC SPECKLE PATTERN INTERFEROMETRY

A short course on Electronic Speckle Pattern Interferometry will be given in two parts: Sunday, 12 May, from 1:00 p.m. to 5:00 p.m. and Monday, 13 May, from 8:00 a.m. to 12:30 p.m. in the Clements Room.

The instructor is Thomas Moore, Professor of physics at Rollins College. Electronic speckle pattern interferometry (ESPI) is an optical method that allows scientists and engineers to visualize the deflection shapes of vibrating objects in real time.

Onsite registration at the meeting will be on a space-available basis.

16. GALLERY OF ACOUSTICS

The Technical Committee on Signal Processing in Acoustics will sponsor the 18th Gallery of Acoustics. 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 will be located in the Grand Ballroom Foyer. Ballots will be distributed to meeting attendees to rank-order the entries. A cash prize of USD $400 and $200 will be awarded to the winning and first runner-up entries.

17. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday at The Galt House. The schedule and rooms for each Committee meeting are given on page A15

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.

18. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 15 May, at 3:30 p.m. in Grand Ballroom B.

ASA scholarship recipients will be introduced. The Rossing Prize in Acoustics Education will be presented to Stanley Chin-Bing. The William and Christine Hartmann Prize in Auditory Neuroscience will be presented to Glenis Long. The Silver Medal in Engineering Acoustics will be presented to

Thomas B. Gabrielson. The R. Bruce Lindsay Award will be presented to Adam Maxwell. The Helmholtz-Rayleigh Medal in Psychological and Physiological Acoustics, Speech Communication, and Architectural Acoustics will be presented to Barbara G. Shinn-Cunningham, and the Gold Medal will be presented to William J. Cavanaugh.

Certificates will be presented to Fellows elected at the Victoria meeting. See page 1831 for a list of fellows.

All attendees are welcome and encouraged to attend. Please join us to honor and congratulate these medalists and other award recipients.

19. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will be held at the Louisville 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 A15 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: Nancy Blair-DeLeon, ASA Standards Manager, Standards Secretariat, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; T.: 631-390-0215; F: 631-923-2875; E: asastds@acousticalsociety.org

20. COFFEE BREAKS

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

21. A/V PREVIEW ROOM

The Stanley Room on the second floor will be set up as an A/V preview room for authors' convenience, and will be available on Sunday from 3:00 p.m. to 5:00 p.m., Monday through Thursday from 7:00 a.m. to 5:00 p.m. and Friday from 7:00 a.m. to 12:00 noon.

22. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Louisville 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 for downloading without charge to anyone in the world. 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. 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://acousticsauthors.org. Published papers from previous meeting can be seen at the site http://asadl/poma.

23. E-MAIL AND INTERNET ZONE

Wi-Fi will be available in all ASA meeting rooms and spaces.

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 on the second floor in the Grand Ballroom foyer.

Tables with power cords will be set up in the Grand Ballroom foyer for attendees to gather and to power-up their electronic devices.

24. SOCIALS

Socials will be held on Tuesday and Thursday evenings, 6:00 p.m. to 7:30 p.m. in Grand Ballroom A/B.

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 open meetings of Technical Committees that begin immediately after the socials

25. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture, sponsored by the College of Fellows, will be held Thursday, 16 May, at 12:00 noon in the Sampson room. The speaker will be Andy Cavatorta who will present the lecture titled "Music, Machines, and Meaning."

This luncheon is open to all attendees and their guests Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 15 May, The cost is USD $30.00 per ticket.

26. 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 on Monday, 13 May, from 5:00 p.m. to 5:30 p.m. in the Willis room. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in the Wilkinson room where refreshments and a cash bar will be available.

The Students' Reception will be held on Wednesday, 15 May, from 6:00 p.m. to 8:00 p.m. in the Wilkinson room . 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.

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.

27. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:45 a.m. on Wednesday, 15 May, in the Sampson room. Those who wish to attend must purchase their tickets in advance by

10:00 a.m. on Tuesday, 14 May. The fee is USD $30 for non-students and USD$ 15 for students.

28. JAM SESSION

You are invited to the JAM on Wednesday night, 15 May, from 8:00 p.m. to midnight (See Mobil App for location). 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.

29. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Louisville meeting. The on-site registration fee for accompanying persons is USD $200. A hospitality room for accompanying persons will be open in the Sampson room, 8:00 a.m. to 10:00 a.m. Monday through Friday. This entitles you access to the accompanying persons room, social events on Tuesday and Thursday, the Jam Session, and the Plenary Session on Wednesday afternoon.

The program will include speakers on the history and culture of the city. Check back to the meeting website for updated information.

Louisville is a city of unique culture. Although bourbon and horse racing in many ways define Louisville, it is also known as a city of compassion, with vibrant arts and food communities. Numerous attractions and a broad range of culinary experiences are within a short distance from The Galt House.

30. WEATHER

May is typically ideal springtime weather in Louisville. Days are warm, but not hot, and nights are cool. On average, daily high temperatures are 77°F, and daily lows are 57°F. Springtime rainfall is not uncommon, so raincoat or umbrella are recommended..

31. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Christin Stilp, Chair; David Knobles, Acoustical Oceanography; Laura Kloepper, Animal Bioacoustics; Shane Kanter, Benjamin Bridgewater, Architectural Acoustics; Kim Kang, Siddhartha Sikdar, Biomedical Acoustics; D. Keith Wilson, Computational Acoustics; Benjamin Tucker, Matthew Kamrath, Education in Acoustics; Michael Haberman, Caleb Sieck, Engineering Acoustics; Whitney Coyle, Peter Rucz, Musical Acoustics; William Murphy, James E. Phillips, Noise; Kevin Lee, Physical Acoustics; Anna Diedesch, Ellen Peng, Psychological and Physiological Acoustics; Lee Culver, Ryan Harne, Signal Processing in Acoustics; Rajka Smiljanic, Speech Communication; Benjamin Shafer, Robert A. Koch, Structural Acoustics and Vibration; Dajun Tang, Underwater Acoustics; Michael Rollins, Kieren Smith Student Council.

32. MEETING ORGANIZING COMMITTEE

Pavel Zahorik, Chair; Christian Stilp, Technical Program Chair; Shae Morgan, Maria Kondaurova, Student Coordinators; Olaf Strelcyk, James Shehorn, Signs; Brett Bachmann, Accompanying Persons Program.

33. PHOTOGRAPHING AND RECORDING

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

34. ABSTRACT ERRATA

This meeting program is Part 2 of the March 2019 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*.

35. GUIDELINES FOR ORAL PRESENTATIONS

Preparation of Visual Aids

- See the guidelines for computer projection in section 41 below.
- 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.
- 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:

- Statement of research problem
- Research methodology
- Review of results
- 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.

36. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS

Content

The poster should be centered around two or three key points supported by the title, figures, and text. The poster should be able to "stand alone." That is, it should be understandable even when you are not present to explain, discuss, and answer questions. This quality is highly desirable since you may not be present the entire time posters are on display, and when you are engaged in discussion with one person, others may want to study the poster without interrupting an ongoing dialogue.

- To meet the "stand alone" criteria, it is suggested that the poster include the following elements, as appropriate:
- Background
- Objective, purpose, or goal
- Hypotheses
- Methodology
- Results (including data, figures, or tables)
- Discussion
- Implications and future research
- References and Acknowledgment
- Design and layout
- A board approximately 8 ft. wide × 4 ft. high will be provided for the display of each poster. Supplies will be available for attaching the poster to the display board. Each board will be marked with an abstract number.
- Typically posters are arranged from left to right and top to bottom. Numbering sections or placing arrows between sections can help guide the viewer through the poster.
- Centered at the top of the poster, include a section with the abstract number, paper title, and author names and affiliations. An institutional logo may be added. Keep the design relatively simple and uncluttered. Avoid glossy paper.

Lettering and text

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

Visuals

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

Presentation

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

Other suggestions

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

37. GUIDELINES FOR USE OF COMPUTER PROJECTION

A PC computer with monaural audio playback capability and 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 must save their presentations for projection in PC format. Also, authors who plan to play audio during their presentations should insure that their sound files are also saved on the CD or USB drive.

Introduction

- 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.

Guidelines

- Set your computer's screen resolution to 1024 x 768 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.

38. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, or to obtain instructions for the preparation and submission of meeting abstracts, contact the Acoustical Society of America, 1305

Walt Whitman Road, Suite 300, Melville, NY 11747-4300; Telephone: 516-576-2360; Fax: 631-923-2875; E-mail: asa@acousticalsociety.org

178th Meeting, San Diego, CA, 30 November–4 December 2019

179th Meeting, Chicago, Illinois, 11–15 May 2020

180th Meeting, Cancun, Mexico, fall 2020

181st Meeting, TBD

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Session 1aAA

Architectural Acoustics, Noise, Signal Processing in Acoustics, and Speech Communication: Integrated Approach to Speech Privacy

Kenneth W. Good, Cochair
Armstrong, 2500 Columbia Ave., Lancaster, PA 17601

Eric L. Reuter, Cochair
Reuter Associates, LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801

Chair's Introduction—9:30

Invited Papers

9:35

1aAA1. Speech privacy measurements and metrics. Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

What is the current state of speech privacy measurements and metrics for built environments? This paper will provide an update on findings from the TCAA Speech Privacy Subcommittee. We will explore the uses, pros, cons, and appropriateness of Speech Privacy Potential (SPP) speech Privacy Index (PI) and Speech Privacy Class (SPC).

9:55

1aAA2. Speech privacy comparison of metrics. Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

How do different speech privacy metrics compare, what are the correlations and differences, and what might we conclude from them? This paper will start with the noise reduction results from several common architectural assemblies. From those results, we will calculate each of the popular speech privacy metrics. Finally, we will explore the correlations and differences of the results.

10:15

1aAA3. A review of speech privacy terms and methodology in building codes, guidelines, and standards. Benjamin Shafer (Tech. Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

Terminology and methodology for the measurement and classification of speech privacy in buildings are globally diverse and cover speech transmission in a relatively broad range of environments from large open spaces to small divided enclosures. This paper is meant to provide a review and summary of terms and methodologies related to speech privacy throughout the global building noise control industry and regulatory organizations. Differences in terminology and speech transmission measurement between standards will be discussed as well as the progressive use of these standards in building codes and guidelines. A more thorough examination of this documentation is intended to provide the building noise control community with both greater insight into the incorporation of speech privacy in regulatory and standard documentation as well as to provide a foundation for greater innovation in building noise control specific to speech transmission.

10:35

1aAA4. Statistical distribution of ambient noise levels compiled from building acoustics measurements. John Loverde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

One of the primary factors affecting speech privacy is the background noise level. During the course of building acoustics testing by the authors' company, the background noise level has been measured in hundreds of locations nationwide in a wide variety of site conditions and building types. This information is compiled and analyzed. The effect of the statistical expectation of the background noise level on the required level of sound reduction is discussed.

10:55

1aAA5. An investigation of the significance of frequency weighting in speech intelligibility calculations. Ric Doedens (K.R. Moeller Assoc. Ltd., 1050 Pachino Court, Burlington, ON L7L 6B9, Canada, rdoedens@logison.com)

Differing approaches to quantifying speech intelligibility exist within ASTM architectural acoustic standards E1130 and E2638. E1130 was written for open office conditions and to cover a wide range of intelligibility levels. E2638 was written for enclosed room scenarios and focuses on speech privacy at confidential levels or higher. Each is currently being evaluated with respect to the possibility of expansion to include both open and closed settings. One difference between the two methodologies pertains to whether background sound level frequencies are weighted or averaged. This paper explores how frequency weighting of background sound levels is significant in the determination of speech intelligibility/privacy/security.

11:15

1aAA6. Privacy in a corporative office. Sergio Beristain (IMA, ESIME, IPN, P.O. Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

A corporative headquarters were installed in three floors of a new ten story building, where all the needed spaces were properly distributed as requested by the customer, which included private offices, meeting rooms, open plan offices, etc., the former two were located in one end of the two highest floors chosen, away from the stairs and elevator. Within weeks after the inauguration of the venue, the company directives noticed that some key design and production information was being leaked somehow and were not able to find out the process, which was a major problem for the company. They requested a quick and efficient solution to this matter, because they could not stop their activities, and they were not willing to lose any more information.

Contributed Paper

11:35

1aAA7. Reverberation analysis and acoustical modeling for improved acoustical conditions within modern workplace phone rooms. Alex Maurer, Jeffrey Fullerton (Intertek, 50 Summer St., Boston, MA 02110, alexander.maurer@intertek.com), and Kimteri Kim (Intertek, New York, NY)

As open office floor plans have grown popular, phone rooms have become important for staff to conduct personal conversations in private. Varying in acoustic and holistic design, these rooms are intended to both isolate the occupant's speech from being heard by others in the office and reduce intrusive noise from disrupting these phone conversations. The acoustical properties within the rooms are often ignored, even though these spaces can have rather reverberant conditions and be subjectively very bothersome and awkward to converse within. Several types of porous absorption can help to control the acoustics of these phone rooms which are commonly implemented today. We have studied a variety of absorptive panel samples, placements, and coverage areas to test the effect of absorption in the phone rooms. We measured the reverberation time and impulse response of the rooms with various acoustical finishes to better understand the effectiveness of porous absorption to control the acoustics of these phone rooms.

Session 1aAO

Acoustical Oceanography: Future Directions in Acoustical Oceanography

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

John A. Colosi, Cochair
Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943

Chair's Introduction—8:30

Invited Papers

8:35

1aAO1. Echoes from the ocean's interior: High-frequency observations of ocean phenomena. Thomas C. Weber, Larry Mayer, Anthony P. Lyons, Scott Loranger, Alexandra M. Padilla, and Elizabeth F. Weidner (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

Recent technological advances in high-frequency (>10 kHz) sonar transducers, sonar transceivers, and sonar design have been accompanied by increased capabilities for observing ocean phenomena. These advances include the high range resolution and frequency-domain target classification capabilities associated with wideband acoustic echo sounders, the long-range high-resolution synoptic imaging capabilities associated with multibeam echo sounders and synthetic aperture sonar, and an increased focus on sensor calibration for all systems. High-frequency sonars are increasingly being used to quantify ocean phenomena at scales ranging from subcentimeter (e.g., individual gas bubbles) to 100s of km (e.g., internal waves) to several 10s of km (e.g., thermohaline staircases). In this talk, we highlight some of the ocean processes that we have been investigating using high-frequency sonar systems, typically involving the transport of hydrocarbons, heat, energy, and fresh water into and through the ocean, and some of the (many) acoustic challenges that must be overcome to continue to increase the value of these observations.

8:55

1aAO2. An advanced sensor platform for acoustic quantification of the ocean twilight zone. Andone C. Lavery, Timothy K. Stanton (Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu), J. Michael Jech (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), and Peter Wiebe (Woods Hole Oceanographic Inst., Woods Hole, MA)

The ocean twilight zone (OTZ) is the vast, globe-spanning, layer of water between 200 and 1000 m depth—home to diverse communities of small fishes, cephalopods, crustaceans, and gelatinous organisms. Yet, little is known about the biology, abundance, biomass, and distribution of these organisms. The OTZ is difficult to sample due to a combination of organism patchiness and avoidance, and difficulties capturing fragile species. Recent evidence suggests that the global OTZ biomass may be sufficient to commercially harvest. Furthermore, much of this biomass performs daily vertical migration (DVM) and may play a potentially critical role in regulating Earth's climate through the export of carbon to the deep ocean. *Deep-See*, an advanced senor platform, was developed to fill the technological void for characterizing the OTZ. This towed vehicle integrates wide-band, split-beam acoustics (1–500 kHz) with optical, environmental, and eDNA sensors that can address many of the challenges associated with sampling in the OTZ. Data from the inaugural cruise in August 2018 highlight that (i) a surprisingly high abundance of organisms can be found outside the dense sound scattering layers and (ii) the target strength of many organisms that perform DVM changes with the depth, which is critical to estimate biomass.

9:15

1aAO3. Sound propagation in the surface mixed layer and upper ocean: An overview of relevant ocean dynamics and mode scattering theory. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

The mixed layer and upper ocean are a region of immense interest to physical oceanographers, meteorologists, and climate scientists because this is the boundary through which energy, momentum, buoyancy, and gasses are exchanged between the ocean and the atmosphere. The upper ocean is also a rich ecosystem for a vast array of ocean organisms and marine wildlife as well as a region of great Naval tactical importance. Given these factors, it is rather surprising that ocean acoustics has paid little attention to this significant region, aside from very high-frequency studies of bubble properties and gas entrainment. From the standpoint of transmission loss, the major work on the problem seems to go back to the Acoustic, Meteorological, Oceanographic Survey (AMOS) of the mid-50s. Here, a review is presented of relevant ocean processes that may be important for acoustic propagation at frequencies ranging from several hundreds of hertz to several kilohertz, where the foundation of the analysis is normal mode and transport theory. Processes of interest are surface gravity waves including subsurface currents, internal waves and tides, wind driven inertial oscillations and Langmuir circulations, eddies and submesoscale processes, turbulence and spice, bubbles, and fishes.

9:35

1aAO4. Ocean remote sensing using passive acoustics. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Conventional acoustic remote sensing techniques typically rely on controlled active sources which can be problematic to deploy and operate over the long term, especially if multiple sources are required to fully illuminate the ocean region of interest. Conversely, receiver arrays are becoming increasingly autonomous, miniaturized, and capable of long term deployment thus enabling passive acoustics for ocean remote sensing applications by taking advantage of uncooperative sources of opportunity (e.g., shipping noise) or the ubiquitous ocean ambient noise which have not typically been used in traditional ocean sensing applications. This fully passive approach can also be advantageous when regulations forbid the use of active sound sources or when no active sources are readily available—e.g., at very low frequencies (~10 Hz) or during covert operations. This presentation will discuss the recent development of ocean remote sensing using passive acoustics notably (1) passive acoustic thermometry to estimate deep ocean temperature variations and internal tides using coherent processing of low-frequency ambient noise; (2) localization of drifting sensor networks using ambient noise to enable random volumetric *ad hoc* receiver array for tracking underwater targets; and (3) monitoring of the shallow water sound channel using shipping sources of opportunity.

9:55

1aAO5. Exploiting ambient noise in polar regions to study ice-ocean interactions. Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St., La Jolla, CA 92093-0238, gdeane@ucsd.edu)

We explore recent developments and future directions for ambient noise cryology: the study of ice-ocean interactions using their underwater noise signatures. The study of ice-ocean interactions is currently spurred by climatic shifts in polar regions and their implications, which include sea level rise and geopolitical stability. There are many important ice-ocean interactions, and a broad range of observational techniques are used to study them, such as satellite remote sensing, ship-based observations with *in situ* sensors and AUV's, boreholes, ground-penetrating radar, photogrammetry, seismometry, and differential GPS. Despite such an extensive suite of techniques, submarine calving and ice melting—which play a critical role in the mass balance of ice shelves and marine-terminating glaciers—remain difficult to measure. Progress in quantifying these processes with their underwater noise signatures will be discussed along with future directions for the field.

10:15–10:30 Break

10:30

1aAO6. Better together—Combining acoustics and environmental DNA to understand ecosystems. Jennifer L. Miksis-Olds (School of Marine Sci. & Ocean Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NC 03824, j.miksisolds@unh.edu) and Alison Watts (Civil and Environ. Eng., Univ. of New Hampshire, Durham, NH)

Acoustic signals have historically been and presently are the state-of-the-art for sensing the ocean at small to large spatial scales. Passive acoustics non-invasively assess sound levels, surface conditions, human activity, and the distribution of vocalizing marine life. Echosounders provide acoustic backscatter information that contribute not only critical information on biology but also physical components of the water column which has supplied invaluable knowledge on the community structure, organism size and distribution, and oceanic microstructure. Advances in DNA methods present an opportunity to harness a new technology and fundamentally improve our capacity to monitor habitats, communities, and individual species. Environmental DNA (eDNA) includes whole microorganisms, tissue fragments, reproductive and waste products, and other cellular material in a sample. eDNA methods, such as acoustics, allow for the identification of species without having to physically capture animals. While eDNA and acoustics are both powerful methods for identifying and describing organisms present in the environment, no single survey method can fully represent the "real" condition. We envision a future where remote platforms can gather and relay data in near-real time and can elicit a targeted response to information, such as launching a drone to survey additional locations or increasing the sampling frequency when key species are present.

Contributed Papers

10:50

1aAO7. Low frequency acoustical scattering from dynamic schools of swim bladder fish. Luis Donoso and Christopher Feuillade (Inst. of Phys., Pontifical Catholic Univ. of Chile, Av. Vicuna Mackenna 4860, Macul, Santiago, Region Metropolitana de Santiago 7820436, Chile, lldonoso@uc.cl)

A time-domain computational model is used to describe the dynamic evolutions of fish schools. The individual and ensemble fish behaviors are governed by three radial parameters, representing attraction, orientation, and repulsion zones between fishes. Different combinations of these radii cause the schools to evolve into various discoid, swarming, parallelized, or toroidal geometric forms. A previously developed model [*J. Acoust. Soc. Am.* **99**, 196–208 (1996)] is applied to study the variations in ensemble scattering from these school types for ensonification frequencies in the swim bladder resonance region, as a function of frequency and time, and predicts distinct characteristic features for the different school geometries. In particular, this work focuses on scattering from disk-shaped schools, by examining the computed resonance response for different school dimensions, packing density, and orientation, in order to identify specific features that are determinative for this type of arrangement. The ultimate goal of this work is to achieve a better understanding of the physical basis for variations in fish school scattering levels and to obtain a detailed statistical description of these objects. [Work supported by ONRG.]

11:05

1aAO8. The measurement of ocean acidity using the depth-dependence of ambient noise. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., P.O. Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca) and Michael J. Buckingham (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA)

The absorption of sound in seawater is due to the viscous and chemical relaxation of different compounds. Over the wind noise band of 1–10 kHz, the frequency dependence of the absorption is due to the mechanisms of

chemical relaxation for magnesium sulfate ($f > 3$ kHz) and for boric acid ($f < 3$ kHz), which involve ionic dissociations activated and deactivated by the condensation and rarefaction of the medium by passing sound waves. Concentrations of both chemicals determine the level at which the sound is absorbed, which makes the process dependent on the salinity, temperature, and pressure in the ocean. The concentration of boric acid in the ocean is a direct measure of pH, while the concentration of magnesium sulfate is independent of pH; thus, a measurement of the frequency dependence of sound absorption may be used to determine ocean acidity. When local winds are strong (> 10 m/s), the ambient noise field is dominated by locally generated surface noise and has a depth-independent directionality and a weakly frequency and depth-dependent intensity, due to sound absorption. By comparing measurements with theory, estimates of ocean acidity can be made from the depth profiles of ambient noise. [Work supported by ONR.]

11:20

1aAO9. Three-dimensional multichannel seismic imaging of water columns in the Gulf of Mexico. Likun Zhang (National Ctr. for Physical Acoust., Univ. of MS, Oxford, MS), Zheguang Zou, Parsa Rad, and Leonardo Macelloni (National Ctr. for Physical Acoust., Univ. of Mississipi, University, MS, zou@olemiss.edu)

Seismic reflection profiling technique, previously used to image the sediments beneath the seafloor, is herein used to image the ocean's water columns, namely, seismic oceanography. The imaging has a much higher lateral resolution (~10 m) than traditional oceanographic measurements such as CTD (usually >100 m). Prior work on seismic oceanography was limited on imaging in two dimensional vertical transects. This work develops the three-dimensional (3D) seismic oceanography technique to image the 3D dynamic processes of water columns. 3D multichannel seismic survey data in a seismic volume of 625 km^3 in the Gulf of Mexico are processed and produce images containing detailed 3D water-column structures near the continental slope. Some mesoscale and sub-mesoscale structures are visualized from different viewing angles. Spectral analyses of the seismic images reveal 3D spatial features of the structures, suggesting the potential of 3D seismic oceanography. [Work supported by NOAA.]

11:35

1aAO10. The soud of light: Towards ocean acoustic sensing with an optical breakdown transducer. Athanasios G. Athanassiadis (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Bldg. 3-257c, Cambridge, MA 02139, thanasi@mit.edu)

When a high-power laser is focused to a small spot in a fluid, nonlinear interactions at the focus can excite a plasma that evolves and glows according to the properties of the breakdown medium. This phenomenon—optical breakdown—is commonly used for underwater chemical measurements in a technique called Laser-Induced Breakdown Spectroscopy (LIBS). However, LIBS generally operates at short ranges and only leverages the optical emission of the plasma for sensing. If instead, the laser systems were tuned to amplify the mechanical effects of optical breakdown, then it could be deployed as a versatile source for broadband acoustic sensing. The optical breakdown acoustic source is simultaneously compact (mm-scale), loud (MPa peak pressures), ultra-broadband (10 kHz–4 MHz), and geometrically reconfigurable. Here, I will describe the physics governing optical breakdown transduction and then show how can this be tuned to enable new single-vehicle sensing strategies in the ocean.

MONDAY MORNING, 13 MAY 2019 NUNN, 8:30 A.M. TO 10:30 A.M.

Session 1aBAa

Biomedical Acoustics: Ultrasound Modeling Workshop

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

A 2-hour hands-on workshop using FOCUS, the 'Fast Object-oriented C++ Ultrasound Simulator' will be held on Monday 13 May, in Session 1aBAa, at 8:30 a.m. in Nunn. Preregistration was required so participation will be on a space-available basis.

Session 1aBAb

Biomedical Acoustics and Signal Processing in Acoustics: Lung Ultrasound and Tissue Stiffness Method I

Xiaoming Zhang, Cochair
Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

Libertario Demi, Cochair
Information Engineering and Computer Science, University of Trento, Via Sommarive, 9, Trento 38123, Italy

Chair's Introduction—11:00

Invited Papers

11:05

1aBAb1. The prognostic role of ultrasonographic air bronchogram in the management of community acquired pneumoniae in children. Riccardo Inchingolo (UOC Pneumologia, Fondazione Policlinico Universitario A. Gemelli IRCCS, Largo Gemelli, 8, Roma 00168, Italy, riccardo.inchingolo@policlinicogemelli.it), Andrea Smargiassi (UOC Pneumologia, Fondazione Policlinico Universitario A. Gemelli IRCCS, Rome, Italy), Roberto Copetti (Pronto Soccorso e Medicina d'Urgenza, Ospedale Civile Latisana, Latisana, Italy), and Gino Soldati (Ecografia Clinica, Lucca, Italy)

Chest ultrasound is a non-invasive method for evaluating children with suspected community-acquired pneumonia (CAP), allowing close follow-up and reduction of ionizing radiation. We studied the prognostic role of the change of ultrasonographic (US) air bronchogram in the management of CAP in terms of rate of complicated CAP, change in empiric antibiotic therapy, relationship to defervescence time, and length of hospitalization. Patients with diagnosis of CAP and radiographic evidence of lung consolidation were prospectively enrolled. The first chest US examination was performed within 12 h from admission and after 48 h. A new grading system (USINCHILD score) based on the presence and features of air bronchogram was adopted. Thirty six patients (mean age of 5 years) were stratified into two groups according to the presence of an increase in at least 1 grade of US score (Δ US grade), expression of an improvement of lung consolidation. The US grade after 48 h < 1 was associated with an increased risk of complicated CAP (OR: 160.88, p-value: 0.0109) and a longer defervescence time (70 h, p-value: 0.0047). Moreover, Δ US grade ≥ 1 was predictive of a short hospitalization (7 days, p-value: 0.0061). USINCHILD score could be an innovative biotechnology tool for the management of pediatric CAP.

11:25

1aBAb2. The ultrasound transmissible lung: Impact of acoustic conditions in flooded lung on therapeutic ultrasound applications for lung tumour treatment. Frank Wolfram and Thomas G. Lesser (Clinic of Thoracic Surgery / Lung Cancer Ctr., SRH Waldklinikum Gera, Strasse d Friedens 122, Gera 07548, Germany, frank.wolfram@srh.de)

For minimal invasive treatment of Lung Cancer and Metastases, thermal ablation is a valuable tool in case of in-operability. Such a radiofrequency (RFA) is widely used despite its noticeable complication rate. The use of non-invasive modalities such as High Intensity Focused Ultrasound (HIFU) would be beneficial in lung, if acoustically transmissible. For such purposes, One Lung Filling (OLF) was developed, which replaces air with the saline content in lung. Suppositious, acoustic conditions in such a saline tissue compound might be different than in solid tissue and impact the HIFU ablation process in lung. Therefore, our work was dedicated to develop a valid acoustic model of lung in flooded conditions and validate it with findings on the HIFU interaction during OLF on preclinical models. Acoustic parameters were determined using a broadband transmission technique which showed atypical but superior conditions for ultrasound transmission through lung. Heat induction was simulated based on the derived parameters in an acoustic-thermal solver using KZK and Penne's bioheat equation (BHTE). Results showed good agreement to the measurement where ablative temperatures are induced in central lung cancer, while lung parenchyma stayed unaffected thermally and due to cavitation up to intensities of 9.500 W cm^{-2} (p— 9.1 MPa). Based on those findings, future directions for clinical application of therapeutic ultrasound during OLF in lung will be discussed.

11:45

1aBAb3. Assessment of human diaphragm function by ultrasounds. Andrea Aliverti (Dipartimento di Elettronica, Informazione e Bioingegneria, Politecnico di Milano, Via G. Colombo, 40, TBMLab, Milano 20133, Italy, andrea.aliverti@polimi.it)

Thoracic ultrasound can provide a non-invasive technique for human diaphragm functional assessment, which can be used as an alternative to traditional, more challenging, and uncomfortable methods, such as transdiaphragmatic pressure measurement, fluoroscopic sniff test, nerve conduction studies, and electromyography. The variables that can be assessed using ultrasounds are (1) the static

measurement of the end-expiratory diaphragm thickness (Tdi), (2) the dynamic evaluation of the ratio of inspiratory to the expiratory diaphragm thicknesses, reported as the thickening ratio or thickening fraction (TF), and (3) the diaphragmatic excursion. The measurements of Tdi and TF are performed by placing a high-frequency linear probe at the level of the zone of apposition, while diaphragm excursion is measured using a curvilinear probe placed in the subcoastal region and recording diaphragm movements in the M-mode. Intra- and inter-observer reliabilities of the measurement of Tdi and TF are high, and ultrasound estimates of Tdi are correlated to direct anatomical measurements. Tdi can be used to monitor the evolution of diaphragm weakness. The reduced values of TF are associated with diaphragmatic paresis. Diaphragm excursion is sensitive to changes in the respiratory pattern, related to diaphragm's volume generating capacity, and can be used to identify diaphragm weakness. In intubated patients, diaphragm excursion is related to weaning outcome.

MONDAY MORNING, 13 MAY 2019 SEGELL, 9:00 A.M. TO 11:55 A.M.

Session 1aNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Acoustics of Healthcare Facilities

Jay Bliefnick, Chair
Architectural Engineering & Construction, University of Nebraska, 1110 S 67th St., Omaha, NE 68182-0816

Chair's Introduction—9:00

Invited Papers

9:05

1aNS1. Quiet time impacts on the neonatal intensive care unit soundscape and patient outcomes. Jonathan R. Weber, Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, jonryanweber@gmail.com), and Ashley Darcy Mahoney (School of Nursing, The George Washington Univ., Washington, DC)

Healthcare is currently transitioning from prioritizing survival to prioritizing patient care with the expectation of survival. In response, current research intends to explore and ultimately identify an optimal hospital environment. Intensive care units are often susceptible to noisier environments resulting from the requirements of urgent care. Administrative interventions such as Quiet Time are a strategy to reduce noise levels without sacrificing patient care. Despite gaining popularity, there is limited published research that rigorously evaluates the effectiveness of Quiet Time from both acoustical and medical perspectives. The presented work includes a longitudinal study of Quiet Time in multiple neonatal intensive care units. Both acoustical and patient physiological measures were taken with the intention of (1) characterizing the relationship between Quiet Time and the measured soundscape, (2) investigating potential relationships between Quiet Time and infant health, and (3) exploring potential relationships between the measured soundscape and infant health. Results including detailed acoustical analysis of traditional and newly developed metrics and statistical models relating both Quiet Time and soundscape to patient physiological response will be presented. Taken as a whole, the research provides insight into the effectiveness of Quiet Time interventions and the relationships between hospital noise and infant health.

9:25

1aNS2. Subjective and objective assessments of pediatric and neonatal hospital soundscapes. Yoshimi Hasegawa and Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska Lincoln, 6185 Walnut St., Omaha, NE 68106-2125, yhasegawa@unomaha.edu)

Existing literature reveals that overall noise levels in pediatric and neonatal hospital units are exceeding acceptable ranges and exhibiting persistent noise level fluctuations over time and location. Previous studies have also revealed insights into the subjective perception of noise and explored existing noise sources and their measured acoustical properties. However, less has been done regarding clear identification of hospital noise sources related to staff annoyance in addition to detailed, informative representations of the acoustical characteristics of hospital environments. This study utilizes two methods of unsupervised learning techniques— factor analysis and clustering analysis— to assess occupant perception alongside detailed noise level measurements. Data were collected in two pediatric and neonatal hospital units to provide informative representations of the existing acoustical environments. The factor analysis results show three inherent noise categories among the various noise sources in the hospitals. The subsequent multiple linear regression demonstrates potential negative impacts of those noise categories on occupants' psychological responses. The clustering analysis results show some relevant parameters for classifying the noise levels into a number of distinct conditions related to the activity level. This study presents new methods for screening hospital acoustical environments and allows us to gain insight into problematic hospital noise and corresponding perception.

9:45

1aNS3. Hospital design features related to patient experience. Kenton Hummel, Erica E. Ryherd, and Jay Bliefnick (Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, hummelkenton@gmail.com)

Noise can be problematic in hospitals due to concerns over impacts on patients and staff members. Noise in particular is problematic and consistently rated low on patient experience surveys nationwide. Furthermore, previous studies have linked hospital noise to negative reactions for patients including reduced sleep and increased incidence of re-hospitalization. These safety risks—coupled with reduced Hospital Value-Based Purchasing program reimbursements due to low patient experience ratings—result in strong incentives for healthcare providers to provide quieter, safer environments. In this talk, we will explore what is known about hospital design features and patient experience, including a particular focus on noise. Results from ongoing and previous studies reveal new insights into how various features of the built environment are related to patient experience and therefore should be considered in the design and renovation of healthcare facilities.

10:05

1aNS4. Evaluating patient and staff perceptions of soundscape conditions within three hospitals. Jay Bliefnick and Erica E. Ryherd (Architectural Eng. & Construction, Univ. of Nebraska, 1110 S 67th St., Omaha, NE 68182-0816, jbliefnick@gmail.com)

Hospitals can present challenging soundscapes due to the continuous activity found within these environments. Routinely, this leads to poor perceptions of acoustical conditions from both patients and staff, such as in nationally reported HCAHPS patient surveys or hospital-administered staff surveys. In fact, it has been found that patient satisfaction of in-room soundscape conditions is highly related to the overall hospital rating and that staff job performance and satisfaction can be negatively affected by the acoustical environments in which they work. This research addressed these issues by collecting acoustical measurements within 38 patient rooms from 11 units of three individual hospitals, comparing results with patient and staff survey information. Data collected included 24-h occupied sound monitoring of patient rooms and nursing stations, unoccupied BNL sound levels from 20 patient rooms, and impulse response measurements of hospital unit hallways. Results from statistical analyses between measured acoustical data and patient/staff surveys will be reported, along with lessons learned applicable across all three hospitals. Taken as a whole, this study provided new insights into patient and staff perceptions of hospital noise, and could ultimately aid in the design process of new hospitals to improve patient and staff satisfaction.

10:25–10:40 Break

Contributed Papers

10:40

1aNS5. Impacts of noise on staff cognitive performance in a hospital emergency department. Khaleela Zaman, Peter Dodds, Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, khaleela25@yahoo.com), and Paul Barach (Wayne State Univ. School of Medicine, Lincoln, MA)

The soundscape of a typical modern hospital emergency department is undoubtedly noisy. Noise-related stress can contribute to human error, adverse medication events, and physician burnout and may negatively affect physicians' mental health and limit clinicians' ability to provide high-quality patient care. Previous studies have revealed average sound pressure levels for hospitals worldwide to be in significant excess of the World Health Organization firm guidelines. However, neither sound pressure level measurements nor loudness evaluations provide enough insight into the problem or potential solutions. Noise sources in hospitals include monitor alarms, overhead paging, echogenic surfaces, trash bins, ring binders, and patient crying out, among others. These sounds can be abrupt, yet not sustained. In order to evaluate the impact of hospital staff distraction and propensity for human error due to noise, the authors have conducted research at a busy, urban emergency department. The effects of various sonic occurrences on the cognitive load and working memory of physicians operating in the emergency department were assessed using binaural augmented acoustic environments as the backdrop for cognitive executive function evaluations. This paper discusses the methods for the binaural augmentation, cognitive testing, and initial results and offers interpretation and potential solutions to address these results.

10:55

1aNS6. Acoustic challenges of senior care facilities. Emily Schilb and Edward Dugger (Edward Dugger + Assoc., 1239 SE Indian St., Ste. 103, Stuart, FL 34997, emily@edplusa.com)

The population of the US, Europe, and Asia is aging, and the rate at which individuals are developing neurodegenerative diseases is also increasing. As a result, we are seeing a rise in the number of senior living facilities, ranging from independent living for active seniors to memory care facilities aiding those suffering from Alzheimer's and Dementia. Often these mixed levels of care are housed in one building, with the result being a dynamic facility with uses not limited to residential units and medical clinics but including fitness centers, pools, theaters, auditoriums, recreation rooms, dining rooms, and bars. With such a diverse range of often acoustically incompatible activities and uses, the challenges in designing these facilities quickly become apparent. During this presentation, we will explore the challenges of integrating residential suites with various amenities and the potential solutions for maintaining acoustic isolation without compromising architectural design or functionality.

11:10

1aNS7. Soundscape in dementia care environment. Arezoo Talebzadeh (Design for Health, OCAD Univ., 8 The Esplanade, Unit 2403, Toronto, ON M5E 0A6, Canada, arezoo.talebzadeh@gmail.com), Ramin Behboudi (Professional Engineers of ON, Toronto, ON, Canada), and Andrea Iaboni (Toronto Rehabilitation Inst., Univ. Health Network, Toronto, ON, Canada)

Noise is an important sensory stimulus in any environment, especially in unfamiliar settings. Noise is impossible to ignore, and any disturbing noise or constant sound can be agitating, disturbing, and confusing for people who cannot escape the environment. People with dementia may already feel disoriented, isolated, and confined inside care facilities; uncontrolled sound can add to their anxiety and distress. Soundscape refers to the human perception of the auditory environment in context; it relies not only on the subjective quality of sound by quantifying the sound level but also the objective quality of the auditory environment based on people's perception. The aim of this study is to describe the soundscape of the Specialized Dementia Unit at the Toronto Rehabilitation Institute, through data collection and observation, and to evaluate the quality of soundscape. Results show that the overall sound level (dB) of the unit is higher than recommendations, and also, the observation study shows that higher sound level not necessarily results in negative atmosphere, such as chaos or agitation. The findings prove a need

for further study on the relation between the sound level and the perception of sound in dementia care units, which can foster improvement in quality of life for residents and staff.

11:25

1aNS8. Experimental study on effect of background noise on deep sleep in bedroom. Xiang Yan (Tsinghua Univ., Rm. 104, Main Academic Bldg., Beijing 100084, China, xiang.yan72@yahoo.com), Jianghua Wang, Hui Li (Beijing Deshang Jingjie Technol. Development Co. Ltd., Beijing, China), and Yuxiao Chen (Tsinghua Univ., Beijing, China)

One of the most important external factors for sleep quality is noise. Previous studies show that deep sleep is disturbed strongly by noise, resulting in insufficient cerebral cortex deep resting, in delaying the growth and development, and in reducing immunity and brain functions. In the past, all the research studies added artificial noise into the bedroom, and how background noise affects sleep was still unknown. In this paper, the difference in the deep sleep length between the normal bedroom and the 0-dB(A) silence room which excludes the background noise during subject's sleep was compared. Continuously two night sleeps of 35 random subjects, wearing the EEG brainwave Zeo headband, were recorded, one in the silence room and another in subject's self-home bedroom. The result shows that comparing 29–35 dBA with the 0 dB extreme silence condition, deep sleep length increased. The best value locates about 31 dBA, which may be related to the masking of breathing sounds. Below 31 dBA, with noise reducing, the deep sleep length is reduced because of the excessive quietness reduces the masking effect, highlighting the effects of incidental noise. Between 31 and 48 dBA, as the noise increases, the length of deep sleep would decrease, by 4% for each additional 1 dB noise.

11:40

1aNS9. Does noise sensitivity or attentional capacity predict cardiovascular responses to distracting sound? Jordan N. Oliver (Purdue Univ., West Lafayette, IN), Lea Sollmann, Annika Niehl (Systems Neurosci. & NeuroTechnol. Unit, HTW Saar, Saarbruecken, Germany), and Alexander L. Francis (Purdue Univ., SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu)

Workplace noise may cause stress that has long-term implications for health. Employees in open-plan offices typically identify intermittently occurring background sounds as significant sources of stress, possibly because they distract attention away from intended tasks. Thus, individual differences in attention and noise sensitivity may explain individual differences in physiological responses to workplace noise. In previous research [Oliver, *et al*., in 176th Meeting of the Acoustical Society of America (2018)], we reported results from a preliminary analysis of 19 participants' affective and electrodermal responses to working in background noise that included intermittent environmental sounds. Electrodermal responses were used as a measure of distraction because they constitute a primary component of the physiological orienting response, a multi-component response reflecting exogenous capture of attention. Our results suggested that individuals who are more sensitive to noise experienced greater frustration with the primary task, but there was no relationship between frustration and distraction as indexed by electrodermal responses. Here, we present additional analyses of an expanded dataset, focusing on attentional and cardiovascular measures (heart rate and pulse volume amplitude) previously linked to cognitive demand and noise annoyance (Francis *et al*., 2016) and relate these results to an overall characterization of the auditory orienting response.

Session 1aPA

Physical Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, Noise, Psychological and Physiological Acoustics, and Speech Communication: Battlefield Acoustics I

W. C. Kirkpatrick Alberts, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20723

Gregory W. Lyons, Cochair
Construction Engineering Research Laboratory, U.S. Army Engineer Research and Development Center, 2902 Newmark Dr., Champaign, IL 61822

Chair's Introduction—8:40

Invited Papers

8:45

1aPA1. Artillery location: Battlefield acoustics in the First World War. R. D. Costley (U.S. Army Engineer Res. and Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, casa.costley@gmail.com)

Although acoustics has played a role in warfare for hundreds, if not thousands, of years, the First World War came after the technological advances of the late 19th and early 20th centuries that enabled more quantifiable and less subjective observations. In 1914, after the outbreak of the war, Frenchman Charles Nordmann, a Professor of Astronomy at the Paris Observatory at Meudon, conceived and developed systems for locating artillery by measuring the differences in the times of arrival of the sound from the artillery to different observation positions. In one system, humans made "subjective" observations; alternatively, an "objective" system recorded these sounds with carbon microphones. In July 1915, Colonel Coote Hedley, head of the Geographical Section of the General Staff in London, learned of Nordmann's work on a visit to France and recruited Second Lieutenant William Lawrence Bragg to evaluate it. Bragg led the allied effort to improve and deploy sound-ranging systems. In developing the system, they encountered problems in sensor design, array design, wind noise, and outdoor sound propagation. The technical achievements gained and obstacles encountered will be described, along with the impact they made to the war effort. Permission to publish was granted by Director, Geotechnical and Structures Laboratory.

9:05

1aPA2. Ambient infrasound noise in urban environments. Sarah McComas (U.S. Army Res. and Development Ctr., 3909 Halls Ferry Rd. ATTN: CEERD-GS-S, Vicksburg, MS 39180, sarah.mccomas@usace.army.mil), Chris Hayward (Roy M. Huffington Dept. of Earth Sci., Southern Methodist Univ., Dallas, TX), Christopher Simpson (U.S. Army Res. and Development Ctr., Vicksburg, MS), Brian W. Stump (Roy M. Huffington Dept. of Earth Sci., Southern Methodist Univ., Dallas, TX), and Mihan H. McKenna (U.S. Army Res. and Development Ctr., Vicksburg, MS)

Associated with the emerging use of infrasound at tactical ranges (less than 50 km in complicated terrain), there is an increased need to deploy infrasound arrays in urban environments near sources of interest in order to record small signals. Initial urban deployment research demonstrated successful monitoring of a single source within 1 km utilizing rooftop infrasound arrays. In order to understand the applicability of this technique for a broader range of sources, the impact of the urban array design on noise reduction and detection of signals of interest must be assessed. These source signals, recorded within an urban scenario, are embedded in a complicated ambient noise environment with multiple coherent clutter sources and an overall decrease in the signal-to-noise ratio. Understanding these ambient acoustic field characteristics requires deconvolving observed coherent signals from time-varying atmospheric effects for different levels of urbanization. This paper presents an initial approach to instrumenting the urban environment with consideration of wind filter effects and data analysis to estimate total ambient acoustic field for a subset of urban zones. This work then provides a definition of urbanization that can be related to the infrasound wavefield. Permission to publish was granted by the Director, Geotechnical and Structures Laboratory.

9:25

1aPA3. Integrating acoustic shooter detection into a hearing protection device. Sébastien Hengy, Marcus Christoph, Pascal Duffner, Pascal Hamery, and Veronique Zimpfer (French-German Res. Inst. of Saint-Louis (ISL), 5 rue du Général Cassagnou - BP 70034, Saint-Louis 68300, France, sebastien.hengy@isl.eu)

Recent advances in the development of hearing protection devices open new fields of applications on the battlefield. While the TCAPS (Tactical Communication And Protective Systems) protect against acute acoustic traumas, they maintain information relative to the acoustic environment of the soldiers. Today, these systems are efficient for both hearing protection and communication. We propose to use the microphones equipping the TCAPS headsets in order to detect and localize shooters on the battlefield. The microphone underneath the hearing protection is used in order to detect the shock and muzzle waves generated by supersonic shots. A meshed network between the TCAPS deployed on the field allows transmitting asynchronous information relative to the detected waves to data fusion nodes that allow estimating the shooter's position. Solutions are proposed in order to compensate the effects of the presence of the head between the microphones underneath the hearing protection. Results concerning the estimation of the time difference of arrival of a transient wave in free field and in the presence of an artificial head are presented. The data fusion process is tested thanks to simulations in various deployment configurations.

9:45

1aPA4. Characterization of digital MEMS microphone elements for usage in a hostile fire detection system on a multi-rotor drone. Wayne E. Prather, William G. Frazier, and Xiao Di (NCPA, Univ. of MS, 145 Hill Dr., P.O. Box 1848, MS 38677-1848, wayne@olemiss.edu)

Acoustic based Hostile Fire Detection Systems (HFDS) depend on multi-element arrays of microphones which are simultaneously sampled at sample rates often in the 10–50 ksps range. A driving component of the electronic hardware complexity of the system is the signal buffering, filtering, and analog to digital conversion required for each microphone channel. Digital MEMS microphones consolidate these functions to within the microphone element giving a digital output representative of the measured acoustic signal. Digital MEMS microphones allow for a completely digital hardware solution which reduces complexity and allows more flexibility in utilizing higher element number arrays if desired. The performance parameters relevant to HFDSs of current best-in-class digital MEMS microphones were measured and compared to current analog microphones commonly utilized in these systems. The purpose of these tests was to determine the viability of digital MEMS microphones as an alternate acoustic sensing element for a HFDS system being implemented on a multi-rotor drone.

10:05–10:20 Break

Contributed Papers

10:20

1aPA5. Overview of acoustic detection for counter unmanned aircraft systems. Brian B. Bishop and Justin Tufariello (The MITRE Corp., 202 Burlington Rd., Bedford, MA 01730, bishopb@mitre.org)

Integration of unmanned aircraft systems (UAS) into the National Airspace System (NAS) is currently a major problem for government and industry. The detection of unauthorized UAS activity is a concern. There is a need for effective low-cost sensor technology to detect and track unauthorized UAS operations. Acoustic sensor systems have potential to improve counter UAS (C-UAS) sensing capabilities at greatly reduced cost relative to traditional sensors such as radar and optical systems, but the commercially available acoustic sensing technology is still lacking. The MITRE Corporation has been a leader in advising the United States' Federal Government on the issues around integration of UAS into the NAS and C-UAS technologies. Research done by MITRE on acoustic sensing of UAS will be presented. The current state of commercially available acoustic sensor systems will be briefly reviewed. Recommendations that MITRE has developed on how to design acoustic sensor systems for C-UAS will be discussed. Future research directions in topics such as acoustic beamforming, array design, and outdoor sound propagation will be recommended. Approved for Public Release; Distribution Unlimited 18-4035

10:35

1aPA6. Experimental studies of sound propagation over vertical and slanted paths in a turbulent atmosphere. Vladimir E. Ostashev, D. Keith Wilson, Matthew Kamrath, Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu), Michael J. White, Michelle E. Swearingen, and Gregory W. Lyons (U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Sound propagation over vertical and slanted paths differs from horizontal paths due to the height dependence of turbulence parameters and atmospheric stratification. Vertical and slanted paths are important for several applications, such as localization of unattended aerial systems (UASs) with ground-based acoustic microphone arrays, detection of ground-based sources with elevated arrays, and the effect of atmospheric turbulence on sonic booms. This paper describes a comprehensive experiment on near-vertical sound propagation conducted over five days in September 2018 at the National Wind Technology Center (NWTC), located near Boulder, CO. The experiment involved the NWTC 135-m meteorological tower with meteorological instruments installed at 17 heights. Nine microphones were located on three horizontal booms attached to the tower at different heights, and a speaker was placed on the ground. Two reference microphones where located near the speaker. About 65% of the time, the speaker transmitted nine or twelve tones. During the remaining time, 6-ms chirps were transmitted. A preliminary analysis of the signal statistics is presented and compared with theoretical predictions.

10:50

1aPA7. Broadband acoustic vector sensing in a forested environment. Sandra L. Collier, Max Denis, John Noble, Latasha Solomon, David Ligon, Deryck D. James, W. C. Kirkpatrick Alberts, Christian G. Reiff, and Leng Sim (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil)

The horizontal acoustic particle velocity and acoustic pressure, concurrent with atmospheric data, were collected during a series of outdoor field tests. Here, we present the effects of a forested environment on acoustic vector sensing for broadband sources. The sources are representative of battlefield signals of interest. The complex distributions are studied in the unsaturated, partially saturated, and fully saturated regimes. Distinct path-dependent, frequency-dependent propagation effects are observed.

11:05

1aPA8. Beamforming and range-migration localization algorithms using the retrieved Green's function. Max Denis (U.S. Army Res. Lab., 1 University Ave., Lowell, MA 01854, max_f_denis@hotmail.com), Sandra L. Collier, John Noble, W. C. Kirkpatrick Alberts, David Ligon, Leng Sim, Deryck D. James, and Christian G. Reiff (U.S. Army Res. Lab., Adelphi, MD)

In this work, the localization of acoustic noise sources and scatterers in an outdoor environment using Green's function retrieval methods is presented. Of particular interest is the implementation of cross-correlation and multidimensional deconvolution into localization algorithms. Conventional beamforming and range-migration algorithms, implemented with Green's function retrieval methods, are compared as localization techniques for an array system. The accuracy of both algorithms to reconstruct the position of targets (noise sources and scatterers) is investigated. Additionally, the added localization enhancements using the retrieved Green's functions are discussed.

11:20

1aPA9. Developing extrapolated pressure level maps from firearms. Reese D. Rasband, Kent L. Gee (BYU Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, r.rasband18@gmail.com), Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH), Caleb M. Wagner (Human Systems Program Office, Air Force Res. Lab., Dayton, OH), and William J. Murphy (Hearing Loss Prevention Team, National Inst. for Occupational Safety and Health, Cincinnati, OH)

This paper describes the development of spatial maps for both peak and A-weighted equivalent sound levels from various firearms, as measured with different spatial resolutions. Two outdoor datasets that adhered to the MIL-STD-1474E weapon noise measurement standard are considered. First is an extensive measurement of the M16A4 rifle at U.S. Marine Corps Base Quantico that included several shooter configurations and a circular measurement arc with 15-deg spacing and 3.67 m radius [R. D. Rasband *et al*., *J. Acoust. Soc. Am.* **143**, 1935 (2018)]. Second is a measurement of many different firearms using an array with 30-deg spacing and 3.0 m radius, performed in Rudyard Michigan [W.J. Murphy *et al*., *J. Acoust. Soc. Am.* **132**, 1905 (2012)]. Using these levels and conservative estimates based on spherical spreading and symmetry, level maps are created based on shooter position, weapon used, and number of shots fired. The fidelity of the extrapolation procedure and possible future improvements is discussed. [Work supported by ONR, WTB Quantico, and AFRL through ORISE.]

Session 1aPP

Psychological and Physiological Acoustics: Physiology and Modeling

Anahita H. Mehta, Cochair
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Axel Ahrens, Cochair
Department of Electrical Engineering, Hearing Systems group, Technical University of Denmark, Ørsteds Plads, Building 352, Kgs. Lyngby 2800, Denmark

Contributed Papers

10:10

1aPP1. The role of cue enhancement and frequency fine-tuning in hearing impaired phone recognition. Ali Abavisani and Jont Allen (ECE, Univ. of Illinois at Urbana-Champaign, 405 N Mathews Ave., Rm. 2137, Urbana, IL 61801, aliabavi@illinois.edu)

A speech-based hearing test is designed to identify the susceptible error-prone phones for individual hearing impaired (HI) ear. Only robust tokens in the experiments' noise levels had been chosen for the test. The noise-robustness of tokens is measured as SNR_{90}s of the token, which is the signal to the speech-weighted noise ratio where an NH listener would recognize the token with an accuracy of 90% on average. Two sets of tokens T_1 and T_2 having the same consonant-vowels but different talkers with distinct SNR_{90}s had been presented at flat gain at listeners' most comfortable level. We studied the effects of frequency fine-tuning of the primary cue by presenting tokens of the same consonant but different vowels with similar SNR_{90}s. Additionally, we investigated the role of changing the intensity of primary cue in HI phone recognition, by presenting tokens from both sets T_1 and T_2. On average, 85% of tokens are improved when we replaced the CV with the same CV but with a more robust talker. Additionally, using CVs with similar SNR_{90}, on average, tokens are improved by 28%, 28%, 25%, and 19%, when we replaced the vowel with ɑ, ɛ, ə, and ɪ. The confusion pattern in each case provides insight into how these changes affect the phone recognition in each HI ear. We propose to prescribe hearing aid amplification tailored to individual HI ears, based on the confusion pattern, the response from cue enhancement, and the response from frequency fine-tuning of the cue.

10:25

1aPP2. Cortical effects on spatial tuning to speech in background babble. Erol J. Ozmeral, Katherine Palandrani, David A. Eddins, and Ann C. Eddins (Commun. Sci. and Disord., Univ. of South Florida, 3802 Spectrum Blvd., Ste. 210, Tampa, FL 33612, eozmeral@usf.edu)

The ability to understand speech in complex backgrounds often relies on spatial factors that contribute to forming discernible auditory objects. From stimulus-evoked onset responses in normal hearing listeners using electroencephalography (EEG), we have shown measurable spatial tuning to moving noise bursts in quiet, revealing a potential window into cortical object formation. However, it is still unknown whether comparable effects are observed with speech stimuli, and whether and how much the presence of noise disrupts EEG responses to moving speech. To test whether the presence of noise has deleterious effects on object formation and potential selective auditory attention, we measured cortical responses to moving speech in the free field with and without background babble (+6 dB SNR) during both passive and active conditions. Active conditions required listeners to respond to the onset of the speech when it occurred at a new location, while indicating yes or no to whether the stimulus occurred at a block-specific location. We discuss in detail the effect of noise and attention on spatial tuning to speech stimuli as measured by the magnitude and latencies of the N1, P2, and P3 components at primary and secondary auditory regions of interest.

10:40

1aPP3. Physiological assays of suprathreshold hearing are consistent with widespread deafferentation of the human auditory periphery. Alexandra Mai, Brooke Flesher, Kelsey Dougherty, Anna Hagedorn, Jennifer M. Simpson, Michael G. Heinz, and Hari M. Bharadwaj (Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, mai10@purdue.edu)

Multiple animal models have robustly shown the effects of noise-exposure and aging can have on the afferent synapses between the cochlea and the auditory nerve. This cochlear synaptopathy can affect responses to suprathreshold stimuli while leaving audiometric thresholds intact. However, currently, there is much debate on whether these same changes occur in humans with significant noise-exposure or with middle age. Our study examined two different physiological responses in which these afferent synapses are a crucial component, the auditory brainstem response (ABR) and the middle ear muscle reflex (MEMR). Versions of these measures were completed in both a clinical setting and a laboratory. Responses to both measures in both testing environments demonstrated significant age and noise-exposure effects. Moreover, these effects remained significant even after statistically accounting for variability in audiometric sensitivity and otoacoustic emissions, suggesting that despite clinically normal audiograms, cochlear synaptopathy may be a widespread occurrence in humans with both acoustic-overexposure and normal aging. Finally, our results suggest that a battery combining ABR and MEMR measures may be viable as a non-invasive assay of synaptopathy and can help examine the perceptual sequelae of such damage.

10:55

1aPP4. Combining psychophysics and modeling to probe suprathreshold auditory processing efficiently. Emmanuel Ponsot (LSP, CNRS/ENS, 1 Pl. Igor Stravinsky, Paris 75004, France, ponsot@ircam.fr)

The ability to understand speech in noise differs considerably among people. Even "normal-hearing" individuals, i.e., with normal audiograms, exhibit a variety of performances in speech-in-noise listening tests. The number of studies investigating the underlying causes of this variability, not captured by current audiological measures, has exploded over the last few years. However, we still lack an overall computational account of suprathreshold auditory processing that would incorporate enough flexibility to capture these idiosyncratic differences. We propose to uncover the psychophysical mechanisms of suprathreshold processing trough its variability among individuals, by combining noise-based psychophysical methods and computational tools. This approach is illustrated in the case of spectrotemporal modulation processing in normal and impaired hearing. Experimental results collected show how listeners modulate their tuning characteristics depending on the task and reveal that hearing-impaired listeners with similar

audiometric losses exhibit a large variety of computational strategies for such tasks, which our modeling approach have the potential to connect with specific computational elements. Overall, this shows how a signal-detection theory framework combined with efficient experimental methods and modeling tools should be considered as a powerful approach to further understand the respective contributions of sensory coding and read-out inefficiencies in human suprathreshold processing.

11:10

1aPP5. Midbrain sensitivity to spectral cues for sound-source localization: A modeling study. Yuxiang Wang and Laurel H. Carney (Elec. Eng., Univ. of Rochester, Rochester, NY, ywang310@ur.rochester.edu)

The features of head-related transfer functions (HRTFs) are a key topic in the field of spatial auditory displays. Neural encoding of HRTFs is not well understood; their sharp spectral peaks and notches are not well represented by average rates of relatively widely tuned auditory-nerve (AN) fibers, and phase-locking to the temporal fine structure is not adequate to encode features at high-frequencies. However, low-frequency fluctuations in cochlear responses to wideband stimuli are encoded in time-varying rates of AN fibers across all frequency channels. Here, we focus on modeling the profile of amplitude fluctuations in response to spectral cues across the population of AN channels. The fluctuation profile across AN responses sets up an average-rate profile across inferior colliculus (IC) neurons, which are sensitive to envelope-related low-frequency fluctuations. Both unilateral and bilateral IC model responses indicate that rather than responding to spectral peaks in a given HRTF, IC responses are more sensitive to fluctuations in frequency channels near steep spectral slopes. The influence of the spatial location and stimulus level on model IC responses was examined. Using statistical methods, psychoacoustical thresholds of discrimination of sounds that differ in the source location were estimated and compared to trends in the existing perceptual results.

11:25

1aPP6. Comparison of the predictive accuracy of different computational models of auditory perception. Evelyn E. Davies-Venn (SLHS, Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55445, venn@umn.edu), Nursadul Mamun (Dept. of Elec. Eng., Univ. of Texas-Dallas, Dallas, TX), Md. Hossain (Faculty of Eng. & IT, The Univ. of Sydney, Sydney, NSW, Australia), Timothy Kwan (Dept. of Biomedical Eng., Univ. of Malaya, Kuala Lumpur, Malaysia), Melanie Putman (SLHS, Univ. of Minnesota, Minneapolis, MN), and M. S. A. Zilany (Dept. of Comput. Eng., Univ. of Hail, KSA, Saudi Arabia)

Computational models of auditory perception offer a time-efficient method of assessing the effects of distortion on speech perception. Several objective metrics have been proposed to predict speech intelligibility, especially when speech is obscured by the presence of background noise. Novel approaches to full-reference and reference-free speech intelligibility metrics have emerged in recent years, but deciphering the best metric for predicting speech intelligibility still requires investigation. This study assessed the predictive accuracy of several reliable, full, and reference-free speech intelligibility metrics. Speech perception scores were measured on listeners with normal hearing and hearing loss in quiet and noise. Acoustic recordings were made of the presented speech stimuli and combined with a computational model of the auditory nerve to simulate behavioral scores using several established metrics such as the STOI, NSIM, SRMR, SII, SNRloss, and BSIM. The estimated speech scores were correlated with behavioral speech recognition scores to assess predictive accuracy of the model simulations. Several of the predicted scores correlated well with behavioral scores. Evaluation of individual phonemes revealed differential sensitivity of the metrics across different phonemic classifications.

11:40

1aPP7. Neural correlates of auditory enhancement. Anahita H. Mehta and Andrew J. Oxenham (Univ. of Minnesota, N640, Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, mehta@umn.edu)

Auditory enhancement is the increase in salience of a target embedded in a simultaneous masker that occurs when a copy of the masker, termed the precursor, is presented first. The effect reflects the general principle of contrast enhancement and may help in the perceptual constancy of speech under varying acoustic conditions. The physiological mechanisms underlying auditory enhancement remain unknown. This study investigated EEG responses under conditions that elicited perceptual enhancement. The target tone was amplitude-modulated at two modulation frequencies to target cortical (~40 Hz) and subcortical (100–200 Hz) responses. Measurements were made in either passive conditions or under active tasks to examine the potential effects of attention on the neural correlates of enhancement. Robust effects of enhancement were observed at the cortical level, replicating our earlier findings. Preliminary data under passive conditions also suggest a trend towards increased neural response to the enhanced target tone at frequencies exceeding 200 Hz, suggesting a subcortical contribution. The results suggest that this paradigm can be used to tap into the neural correlates of auditory enhancement at both cortical and subcortical levels simultaneously and show the potential for tapping into attentional modulation of auditory enhancement. [Work supported by NIH grant R01DC012262.]

Session 1aSAa

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Architectural Acoustics: Smart Materials for Acoustics and Vibration I

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Bogdan Ioan Popa, Cochair
Mechanical Engineering, University of Michigan, University of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Invited Papers

8:00

1aSAa1. Design and additive manufacturing of electro-mechanical metamaterials with designed anisotropy and self-sensing. Xiaoyu (Rayne) Zheng (Mech. Eng., Virginia Tech, 635 Prices Fork Rd., 445 Goodwin Hall, Blacksburg, VA 24061, raynexzheng@vt.edu)

In this talk, I will present our research on design, additive manufacturing, and performances of new classes of multi-functional materials that transcend the common electro-mechanical coupling limitations. These materials are as light as carbon aerogels, but with orders of magnitude higher stiffness and strength, they possess multi-functionalities. They are composed of interconnected 3D hierarchical micro-structures as designed "atoms" and "molecules" as in natural materials to reach uncharted white space in the material selection charts. Attention is focused on our development of a suite of novel additive manufacturing and processing techniques to synthesis traditionally unprocessable inorganic/organic highly responsive building blocks and architect them into scalable form factors with precisely defined active three-dimensional micro- and nano-scale features. We will discuss the possiblities of coupling and decoupling of the piezoelectric coefficients, density stiffness scaling, and selective electro-mechanical amplifications through our metamaterial design and printing approach. These insights shed light on a next generation of metamaterials, with designed-in structural and smart functionalities, including self-sensing and actuation, vector sensing and detection as well as simultaneous impact wave absorptions and self-monitoring with only a fraction of solid, as opposed to relying on multiple components.

8:20

1aSAa2. Dynamic reconfiguration of two-dimensional lattices with bistable springs. Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, julien.meaud@me.gatech.edu)

This study focuses on wave propagation and reconfiguration in two-dimensional square mass-spring lattices that include bistable springs. Due to the presence of directional and/or omnidirectional bandgaps in stable deformed configurations in which some of the bistable springs are in a deformed equilibrium, these lattices have the ability to serve as reconfigurable wave filters and wave guides. In this work, we study non-linear wave propagation in response to stimulus of large amplitude. Mechanical stimuli of large amplitudes have the capability of causing a dynamic reconfiguration of the lattice to a configuration of lower potential energy, which dramatically affect wave propagation. Using numerical simulations, the influence of the stimulus amplitude, duration, and location of the applied stimulus on the reconfiguration is analyzed. The ability to alter in a predictable manner wave propagation in these lattices offers new opportunities for tunable phononic crystals.

8:40

1aSAa3. Optimal vibration suppression in adaptable acoustic metamaterials by a complex wavenumber. Aaron J. Stearns (Mech. Eng., Appl. Res. Lab., Penn State Univ., P.O. Box 30 Burrowes St., State College, PA 16804-0030, ajs6037@psu.edu) and Benjamin Beck (Acoust., Appl. Res. Lab., Penn State Univ., State College, PA)

Acoustic metamaterials are composite materials exhibiting effective properties and acoustic behavior not found in traditional materials. Through periodic subwavelength resonant inclusions, acoustic metamaterials enable steering, cloaking, lensing, and frequency band control of acoustic waves. A common drawback of acoustic metamaterials is that the properties are limited to narrow frequency bands. Investigation of practical active and adaptable acoustic metamaterials is valuable in achieving wider operation frequency bands. Here, a one dimensional metamaterial is analyzed using a finite element method. The purpose of the metamaterial is to minimize vibration. From the finite element method formulation, a complex wavenumber is calculated. The unit cell considered is active and adaptable via piezoelectric actuators attached to negative capacitance shunt circuits. Therefore, the stiffness of the unit cell is tunable through selection of the circuit parameters. Choosing the negative capacitance shunt circuit elements to maximize the attenuative part of the complex wavenumber leads to maximum vibration suppression. So far, it has been found that maximizing the attenuative part of the wavenumber gives unrealizable specifications for the shunt circuit. To prevent this, the goal of this work is to constrain the optimization problem using experimentally obtained circuit stability bounds. The experimental stability results and optimization results will be presented.

9:00–9:15 Break

Contributed Papers

9:15

1aSAa4. Reconfigurable valley acoustic phononic crystals based on a fluidic system. Chen Shen (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC 27708, chen.shen4@duke.edu), Zhenhua Tian (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), Junfei Li (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC), Eric Reit, Tony Jun Huang (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), and Steven Cummer (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC)

Valley serves as a new degree of freedom in controlling wave dynamics. Here, we present a design of valley acoustic phononic crystals (PCs) composed of a hybrid channel-cavity structure. Valley states for both waveguide and surface acoustic modes can be realized, and the mode transition is enabled by adjusting the channel height. Reconfigurable valley Hall phase transition in a wide range of frequencies is allowed by tuning the cavity sizes based on a fluidic system. By injecting/withdrawing fluid into/out of the cavities, the dispersion relation, phase transition, and edge states can be controlled conveniently in the two-dimensional PCs. Frequency-dependent acoustic routing, tunable refraction, topological switching, wave splitting, and reconfigurable acoustic pathways with suppressed backscattering are demonstrated both numerically and experimentally through acoustic field scanning. The reconfigurable valley PCs can serve as a versatile platform for exploring valley-related physics and achieving tunable wideband acoustic devices.

9:30

1aSAa5. Anomalous wave polarization through a 3D periodic auxetic lattice. Ganesh U. Patil and Kathryn H. Matlack (Mech. Sci. and Eng., Univ. of Illinois Urbana-Champaign, 1206 W. Green St., Urbana, IL 61801, gupatil2@illinois.edu)

Solid media supports both longitudinal and shear wave polarizations, providing a rich platform for designing phononic materials with prescribed wave filtering, engineered mode conversions, negative refraction, and other unique properties. While longitudinal waves almost always propagate at a faster velocity than shear waves in natural materials, tailoring the polarization of the faster wave velocity could enable unique control over wave propagation properties. Here, we present a three-dimensional periodic "bowtie" lattice that exhibits a shift in the faster wave polarization from the quasi-longitudinal to quasi-transverse wave in an anisotropic plane. We observe that this shift, termed "anomalous wave polarization", is possible when the lattice behaves auxetically. Using the finite element method, we show that the wave polarizations in the bowtie lattice depend on certain geometric parameters and the wave propagation direction. We use wave velocity information to evaluate the lattice effective properties using the Bloch-wave homogenization approach and confirm the required elastic condition for the polarization anomaly in an anisotropic plane. We also emphasize the importance of identifying the wave polarization along with the wave velocity classification for lattice effective property evaluation. This lattice behavior will be useful for designing mode-converting metamaterials that have potential application in non-destructive testing and bio-medical ultrasounds.

Session 1aSAb

Structural Acoustics and Vibration and Architectural Acoustics: Vibration Reduction for Extraordinarily Sensitive Applications

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Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608

Chair's Introduction—9:10

Invited Papers

9:15

1aSAb1. Ultra-low frequency vertical vibration isolators for absolute gravimeters. Kang Wu, Jiamin Yao, Guan Wang, Gang Li, and Lijun Wang (Precision Instruments, Tsinghua Univ., Bldg. 9003, Qinghuayuan 1, Haidian District, Beijing 100084, China, kangwu@mail.tsinghua.edu.cn)

High-precision absolute determinations of gravitational acceleration g provide important data for many fields such as metrology, geophysics, and geological exploration. In absolute gravimetry, vibrational noise from seismic and other environmental disturbances is one of the limiting factors. Several types of ultra-low vertical vibration isolators have been developed in Tsinghua University. The first one is a passive isolator based on LaCoste spring linkage and can achieve a natural period up to 32 s. The second one is an active isolator employing a two-stage beam structure. The upper beam is suspended from the frame with a hex spring, and the lower beam is suspended from the upper one using a zero-length spring. A feedback circuit is equipped to keep the angle between the two beams at a fixed value. The isolator can achieve a natural period of 100 s. The last one is an active isolator based on a two-stage structure, in which geometric anti-springs are used to support the proof mass. The volume of the isolator is greatly decreased, and the allowable load is increased while maintaining a natural period more than 15 s.

9:40

1aSAb2. Listening to the songs of the universe: How vibration control for the laser interferometer gravitational-wave observatory (LIGO) allows us to measure ripples in the fabric of space. Brian Lantz (Ginzton Lab, Stanford Univ., Spilker Bldg., 348 Via Pueblo Mall, Stanford, CA 94305, BLantz@stanford.edu)

In 2015, humanity made the first detection of gravitational waves from the violent collision of two black holes. As Einstein predicted, this collision sent waves through the fabric of space-time, but nearly 100 years passed between Einstein's prediction and the first measurement of these waves by the advanced LIGO detectors. The detection was made possible by many advances in the precision measurement. I will describe the detectors and one of the key technologies, the vibration isolation for the optics; at 10 Hz, the motion of the LIGO mirrors is at least 1,000,000,000 times less than the motion of the ground. By creating one of the quietest places on Earth, we have created a new way to listen to the stars.

10:05

1aSAb3. An overview of isolation controls at LIGO. Arnaud Pele (Caltech, 19100 LIGO Ln., Livingston, LA 70754, apele@ligo-la.caltech.edu)

The Laser Interferometer Gravitational-Wave Observatory (LIGO) is a large-scale physics experiment that aims at measuring gravitational waves emitted by astrophysical sources. The detection of black hole mergers and neutron star collision was made possible by the extreme level of isolation required to hold the optics still from external ground disturbances in a large band of the spectrum. From the low-frequencies (earthquakes, wind, microseism, below 1 Hz) to the higher frequencies (anthropogenic noise, above 1 Hz), the controls system must be tuned to meet the requirements for lock acquisition, lock stability, and sensitivity of the instrument. In this talk, I will describe the overall control scheme of the LIGO isolation platforms and mirror suspensions, and the challenges met to design the many feedback and feedforward control loops.

10:30–10:45 Break

10:45

1aSAb4. Seismic isolation in advanced Virgo gravitational wave detector. Valerio Boschi (European Gravitational Observatory, via Amaldi, Cascina (PI) 56021, Italy, valerio.boschi@ligo.org)

We will present an overview of the seismic isolation systems used in an AdVirgo gravitational wave interferometer. We will concentrate on the the so-called super-attenuator, the seismic isolator used for all the detector main optical components. This complex mechanical device is able to provide more than 12 orders of magnitude of attenuation above a few Hz. We will also describe its high-performance digital control system and the control algorithms implemented with it.

11:10

1aSAb5. Vibration reduction for gravitational wave detectors. Carlos Frajuca (Mech., IFSP, Rua Medina, 82, Carapicuiba, SP 06355140, Brazil, frajuca@gmail.com), Fabio D. Bortoli (Mech., IFSP, Sao Paulo, SP, Brazil), and Nadja S. Magalhaes (Phys., Unifesp, Carapicuiba, Brazil)

ultimately vibration detectors. They transform the vibrations in space-time into electrical vibrations in the electronic components of the detector. In this work, it is shown that how undesirable vibration is reduced in some of these detectors, even when it is necessary, in order to increase the general sensitivity, to incorporate in such detectors devices that vibrate alone and may compromise the overall performance.

MONDAY MORNING, 13 MAY 2019 BECKHAM, 9:00 A.M. TO 11:35 A.M.

Session 1aSP

Signal Processing in Acoustics, Engineering Acoustics, Physical Acoustics, Underwater Acoustics, and Structural Acoustics and Vibration: Reconfigurable Arrays for Adaptive Wave Guiding

Ryan L. Harne, Cochair
Mechanical and Aerospace Engineering, The Ohio State University, 201 W 19th Ave., E540 Scott Lab, Columbus, OH 43210

Jeffrey S. Rogers, Cochair
Acoustics Division, Naval Research Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375

Invited Papers

9:00

1aSP1. Topology optimization of origami-inspired reconfigurable frequency selective surfaces. Kazuko Fuchi (Univ. of Dayton Res. Inst., 300 College Park, Dayton, OH 45469, kfuchi1@udayton.edu), Andrew Gillman (UES, Inc., Beavercreek, OH), Philip Buskohl (Mater. and Manufacturing Directorate, Air Force Res. Lab., WPAFB, OH), and Alexander Pankonien (Aerosp. Systems Directorate, Air Force Res. Lab., WPAFB, OH)

Fold-driven reconfigurable devices have a potential to expand functional spaces beyond traditional adaptive wave propagation strategies. In particular, designs inspired by the art of origami leverage the mathematics of origami to map designs defined on two-dimensional surfaces to complex three-dimensional shapes. The design space in this paradigm is vast, so a systematic method is needed to design a device that achieves its goal. In our initial effort, we surveyed electromagnetic wave propagation properties of foldable frequency selective surfaces (FSS) and foldable and deployable antennas based on various known origami designs to identify a number of working principles of functional tuning. We incorporated these findings in the implementation of a design method that finds an origami FSS pattern that achieves the desired frequency tuning. This method is adopted from density-based topology optimization, with a notion that anything functional could be described through a distribution of an effective density of the relevant material property. A substrate is "patterned" with foldable segments parameterized through torsional springs; electromagnetically relevant conductive patterns are described as predefined surfaces that remain unchanged. This talk will discuss the lessons learned from our investigations and remaining challenges of designing fold-driven reconfigurable devices for wave propagation control.

9:20

1aSP2. Partial activation of modular and foldable tessellated acoustic arrays for wave focusing. Ningxiner Zhao and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., Columbus, OH 43210, zhao.2684@osu.edu)

Tessellated acoustic arrays inspired by origami structures are suggested to enable wave focusing by exploiting curvatures realized by folded configurations of the array transducer elements. The use of origami-inspired folding patterns also cultivates great portability for space-limited applications. Yet, maintaining curvatures may prohibit feasible implementation of a tessellated been studied for the acoustic energy focusing usage. This research proposes an alternative technique to achieve wave focusing with tessellated arrays that do not realize curvatures upon folding. Here, the partial activation of a tessellated array is exploited to result in constructive interference that realizes a nearfield focal region. The modeling approach to examine partially activated arrays is presented and verified against numerical simulations. Then, the changes in nearfield focusing characteristics are correlated with corresponding changes in partial activation and array folding extent. The opportunities to exploit the partial activation of relatively simple origami-inspired array structures are examined to identify strategies to simplify implementation of portable, folding acoustic arrays in applications.

9:40

1aSP3. A performance metric for screen selection with the acoustic single pixel imager. Juan Ramirez (U.S. Air Force Res. Lab., 329 East 1st St., Apt 306, Dayton, OH 45402, juan.ramirez.ee@gmail.com), Jeffrey S. Rogers, and Geoffrey F. Edelmann (Naval Res. Lab., Washington, DC)

In the recent literature, an Acoustic Single-Pixel Imager has been successfully developed for source localization in a two-dimensional waveguide. Source bearing angle estimation was carried out by applying sparse recovery techniques on sensor measurements taken over dierent imaging screens. In this paper, we show that the mutual coherence of the sensing matrix can be used as a metric to predict the source localization capability of the single-pixel imaging system. In particular, our analysis focuses on the sparsity of open cells within the imaging screen and the number of imaging screens used to maximize the probability of correct detection over varying levels of source sparsity. In this work, we develop a simulation environment to demonstrate how the mutual coherence of the sensing matrix correlates with source localization performance over source sparsity, sparsity of open screen cells, and number of measurements used for sparse recovery. Our analysis shows that the leading factor in source localization performance gains is primarily from the number of imaging screens used to measure the acoustic wave-field.

10:00

1aSP4. Mechanics and dynamics of reconfigurable curved creased origami arrays. Evgueni T. Filipov and Steven R. Woodruff (Dept. of Civil and Environ. Eng., Univ. of Michigan, 2350 Hayward, 2144 GG Brown Bldg., Ann Arbor, MI 48109, filipov@umich.edu)

The principles of origami have allowed for novel deployable and reconfigurable structures with applications from micro-robotics to adaptable architecture. Most origami patterns use rigorous mathematical definitions to achieve their desired folding kinematics; however, these geometries result in discrete and segmented structures that have sharp edges. In contrast, curved creased origami uses a more arbitrary placement of folds and can thus achieve smooth surfaces. These smooth surfaces can be useful for problems in acoustics, fluid flow, electromagnetics, wave-propagation, and more. In this talk, we first explore the geometry of the curved crease origami, where continuous curvatures and elastic bending occur over the entire surface of the thin sheet. We use a simplified bar and hinge model to approximate the folding sequence and model the behaviors of the curved crease origami. The model captures stretching and shearing of the origami, bending along principle curvature directions, and bending at the prescribed curved creases. We explore the stiffness, large deformation mechanics, buckling, and dynamic behaviors of the curved crease origami. Our results show that the curved folding can enable highly tuneable properties for these novel thin-sheet arrays.

10:20–10:40 Break

10:40

1aSP5. Deployable tessellated acoustic array with a curved Miura-ori pattern for ultrasound focusing in multilayered media. Chengzhe Zou and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., Columbus, OH 43210, zou.258@osu.edu)

High intensity focused ultrasound (HIFU) has been successfully applied to treat cancers in clinical settings. In such treatment, the ultrasonic transducer projects focused ultrasound to the diseased tissues for thermal ablation. Nevertheless, the absorption, diffusion, and reflection that occur on the propagation path of ultrasound may reduce the effectiveness of HIFU. In order to overcome this challenge, a concept of origami-inspired deployable tessellated acoustic arrays may be leveraged. Fueled by large portability, the array may be compacted for insertion to the body and guidance to the point of care where the deployed array is then used for treatment. Such a vision requires understanding how wave propagation behaviors from reconfigurable tessellated transducer arrays are tailored in a multilayer environment. Here, the curved Miura-ori tessellation is used to approximate the arc shape for focusing. An analytical modeling framework is extended to investigate the new wave propagation behaviors encountered in biological-like media. Using the analytical tool, the tessellated acoustic array is compared with the ideal arc transducer, and the results indicate that the proposed concept may be comparable with an ideal case in focusing. In addition, the deployability of the tessellated acoustic array is confirmed through experimental efforts in a controlled multilayer environment.

11:00

1aSP6. Experimental observation of valley-Hall edge states in elastic waveguides based on diatomic-graphene-like phononic crystals. Hongfei Zhu (Univ. of Notre Dame, Notre Dame, IN), Ting-Wei Liu, and Fabio Semperlotti (Purdue Univ., 177 S Russell St., West Lafayette, IN 47907, liu2041@purdue.edu)

Inspired by recent discoveries of topological phases of matter in quantum physics, there has been a rapidly growing research effort in creating their analogs in other classical wave systems, including acoustics. Achieving robust wave transmission even in the presence of disorder and defects could have a profound impact on many practical applications and devices. In this study, we report on the design and experimental validation of a fully continuous and load-bearing phononic structural waveguide capable of one-directional guided modes along the walls of topologically distinct domains. The lattice structure of the waveguide is inspired by diatomic graphene which allows realizing an elastodynamic analog of the quantum valley Hall effect (QVHE). Despite the fact that individual bulk properties are topologically trivial (i.e., associated with a zero Chern number), the dynamic behavior acquires topological significance in the neighborhood of the high symmetry points in momentum space; the so-called valleys. Our theoretical and experimental results confirm the existence of protected edge states traveling along the walls of domains having broken space inversion symmetry.

Contributed Paper

11:20

1aSP7. Continuous scan beamforming using a rotating microphone array for mapping of acoustic sources in a soundproof chamber. Abe H. Lee, Andrew White, and Parthiv Shah (ATA Eng., Inc., 13290 Evening Creek Dr. South, Ste. 250, San Diego, CA 92128, abe.lee@ata-e.com)

Continuous scan beamforming (CSBF) is a novel approach that can improve the dynamic range of a microphone array used for source localization. In the conventional beamforming approach in which spatially fixed sensors are used, the number of sensors employed determines the dynamic range of the array. Whereas, in the CSBF approach, by employing moving sensors in a prescribed motion, the effective number of sensors (so-called virtual sensors) used for the beamforming process can be greatly increased, and therefore, it can provide enhanced dynamic ranges close to the theoretical limit. In the CSBF process, for reconstruction of the time data acquired by moving sensors, stationary microphones are used for phase referencing. At ATA Engineering, Inc., a rotating, planar configuration array of 60 microphones was built and tested in a soundproof chamber with spatially distributed acoustic sources inside. The results showed that CSBF has much better performance than the conventional beamforming that employs fixed sensors; CSBF is able to discriminate tested sources that are almost 20 dB apart, which is not possible with the conventional approach. Due to the enhanced dynamic range, CSBF provides much cleaner mapping of the distribution of sources without physically increasing the number of sensors.

MONDAY AFTERNOON, 13 MAY 2019 WILLIS, 1:00 P.M. TO 3:30 P.M.

Session 1pAO

Acoustical Oceanography: Topics in Acoustical Oceanography

Gopu R. Potty, Cochair
Dept. of Ocean Engineering, University of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882

Kevin M. Lee, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Contributed Papers

1:00

1pAO1. Variations of acoustic noise intensity accompanying internal wave solitons. Boris Katsnelson (Marine Geosci., Univ. of Haifa, Mt. Carmel, Haifa 3498839, Israel, bkatsnels@univ.haifa.ac.il), Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA), and Qianchu Zhang (Marine Geosci., Univ. of Haifa, Haifa, Israel)

In this paper, acoustical noise intensity fluctuations recorded by single hydrophones (SHRUs) in the Shallow Water 2006 experiment are studied. The area of experiment (New Jersey Atlantic shelf) is characterized by a remarkable activity of internal waves, in particular, approximately twice per day trains of nonlinear internal waves (NIW) consisting of up to ten separate peaks with the amplitudes about 10–15 m and wave front parallel to the coastal line move toward the beach. During its motion trains of NIW cross positions of five SHRUs, located about 5–8 km to each other along line perpendicular to the coast as well as thermistor's strings which allow us to estimate the shape and evolution of trains while they are propagating. It is shown that the appearance of irregular wideband sound field (amplitude by up to 20–40 dB greater than noise background) takes place when the train of NIW is passing through the location of the corresponding SHRUs. The nature of these signals is discussed, and the spectrum and specific temporal variations as well as other characteristics are analyzed. [Work was supported by ONRG, NSF, and BSF.]

1:15

1pAO2. Variability of the sound field in the presence of internal Kelvin waves in a stratified lake: The Sea of Galilee as a case study. Ernest Uzhansky, Boris Katsnelson (Marine Geosci., Univ. of Haifa, 199 Abba Khouchy Ave., Haifa 3498838, Israel, ericheg@inbox.ru), Andrey Lunkov (Prokhorov General Phys. Inst., Moscow, Russian Federation), and Ilia Ostrovsky (IOLR, I.Allon Kinneret Lab, Migdal, Israel)

The spatiotemporal variability of low- and mid-frequency sound field in the presence of internal Kelvin waves (IKWs) was studied in the Sea of Galilee. Experimental measurements of the sound field were carried out using a vertical line array (VLA) consisting of ten hydrophones with 3 m spacing. The VLA was deployed in the deepest (37 m) part of the lake. Signals were transmitted from the source deployed at the peripheral lake location at a distance of 5.5 km from the VLA at 8-m depth. Linear frequency modulation pulses (300–2000 Hz) were transmitted with 5 sec intervals during >24 h (the period of the IKWs). IKWs were registered using three thermistor chains (TCs) positioned along an offshore transect at 10-m, 20-m, and 37-m station depths. This setting allowed us to characterize the variations of the thermal structure and the corresponding sound speed profile along transect. The vertical structure of the sound field registered with the VLA shows connection with temporal variability of IKWs. The modeling of sound propagation was done using a Parabolic Equation (PE) method, taking into account the parameters of bottom and lake bathymetry. The PE results showed close agreement with our experimental measurements. [Work supported by Israel Science Foundation.]

1:30

1pAO3. Using ambient noise to evaluate the acoustic Green's function for a passive geoacoustic inversion in a dynamic shallow-water environment. Tsu Wei Tan, Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, ttan1@nps.edu), Boris Katsnelson, and Marina Yarina (Univ. of Haifa, Haifa, Israel)

Cross-correlation functions (CCFs) of ambient and shipping noise recorded by two hydrophones approximate the deterministic Green's function and contain information about the propagation environment. This paper employs the data collected in the Shallow Water 2006 experiment on the New Jersey continental shelf to investigate the factors that affect the emergence of approximate Green's functions from noise CCF estimates and accuracy of the approximation. One month-long continuous records of noise obtained by moored single-hydrophone receivers are analyzed. Hydrophones are located in 80–100-m deep water at distances of several kilometers from each other. Rapid variations of the water sound speed profile, which are primarily due to propagation of trains of strong nonlinear internal gravity waves, limit useful noise-averaging time. Available water temperature data are used to guide the selection of time windows for noise averaging and improve CCF evaluation and retrieval of information on seafloor properties. Various approaches to coherent stacking of CCFs are compared. Time warping transform is applied to the resultant noise CCF to extract dispersion curves of acoustic normal modes. The results of the geoacoustic inversion based on the passively measured dispersion curves are compared with the earlier results obtained using controlled sound sources. [Work supported by NSF and BSF.]

1:45

1pAO4. Implementation of random forest in geo-acoustic study. Zhengyu Hou (CAS Key Lab. of Ocean and Marginal Sea Geology, South China Sea Inst. of Oceanology, Chinese Acad. of Sci., 164 Xingang West Rd., Haizhu District, Guangzhou 510301, China, zyhou@scsio.ac.cn)

The correlation between sediment sound velocity (V) and physical properties has been studied for 60 years using empirical formulas and found to be difficult to predict V accurately. Random forest (RF) is a scientific discipline and a method of data analysis that automates analytical model building. Here, we present the implementation of the RF algorithm in V prediction and sediment classification. The goal of this study is to establish a predictive model based on RF using multiple physical parameters (mean grain size, porosity, wet bulk density, and water content). Compared to empirical formulas, the average error of RF velocity is only 0.95%, ranging from 0.03% to 2.73%, which has improved the accuracy of V prediction. We also used Mean Decrease Impurity importance to evaluate the importance of a variable and found that the most important feature in the predictive model is the mean grain size. The classification model based on RF reaching up to 75% accuracy in the dataset. Multiple features, such as physical properties, sedimentary environment, and sediment source, affect the geo-acoustic properties of sediments. The next goal is to use multiple features to improve the model and further improve the accuracy of sound velocity prediction and sediment classification.

2:00–2:15 Break

2:15

1pAO5. The contribution of 12 kHz multibeam sonar to a southern California marine soundscape. Hilary Kates Varghese (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, hkatesvarghese@ccom.unh.edu), Michael J. Smith (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Dover, NH), Jennifer L. Miksis-Olds (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NC), and Larry Mayer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

An ocean mapping survey was conducted over the Southern California Antisubmarine Warfare Range, a hydrophone range to characterize the radiation pattern of the *R*/*V* Sally Ride's EM 122 (12 kHz) Kongsberg multibeam echosounder. Spanning a 2000 km^2 area, the 89-hydrophones in the range, combined with the mapping survey, provided the opportunity to study the contribution of this anthropogenic noise to the marine soundscape. The soundscape was characterized and compared at selected hydrophones across the range before, during, and after the multibeam survey. One minute averages of the sound level were calculated over the data collection period. Sound level percentiles (P1, P10, P50, P90, and P99) were calculated for the full spectrum (1 Hz–48 kHz) and select frequency bands, and spectral probability density plots were generated for each time period. Frequency correlation matrices for each time period were produced and compared using difference matrices to identify changes in the soundscape. The results are placed in the context of the auditory scene of Cuvier's beaked whales resident on the range by applying a mid-frequency marine mammal weighting function. [Work supported by NOAA, ONR, and Scripps Institute of Oceanography.]

2:30

1pAO6. Estimation of environmental parameters with machine learning using a compact tetrahedral array and sources of opportunity. Jesse T. Moore, James H. Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Middleton 14, Narragansett, RI 02882, jesse_moore@my.uri.edu), Aditi Tripathy (Ocean Eng., Univ. of Rhode Island, Kingston, RI), Makio Tazawa (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Jennifer Amaral, Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI), Arthur Newhall, and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

In a previous paper, we showed that we could localize sound sources using a compact tetrahedral hydrophone array in a continental shelf environment south of Block Island, Rhode Island. The tetrahedral array of phones, 0.5 m on a side, was deployed to monitor the construction and operation of the first offshore wind farm in the United States. Directions of arrival (DOAs) for a number of ships were computed using a time difference of arrival technique. Given the DOAs, ranges are estimated using supervised machine learning techniques. Here, we extend this work to estimate a number of environmental parameters including water depth and sediment composition. Training sets of range-dependent ocean waveguides and sediment sound speeds were generated using a propagation model for a neural network. Data from the tetrahedral array were processed by the neural network, which provided estimates of the water depth and sediment parameters such as sound speed and density. These estimates are compared to bathymetric data and core data collected as part of the site characterization for the wind farm. [Work supported by the Office of Naval Research and the Bureau of Ocean Energy Management.]

2:45

1pAO7. Estimation of shipping noise from sparse measurements via generative adversarial networks. Johnny L. Chen (Appl. Res. in Acoust., LLC, 209 N. Commerce St., Ste. 300, Culpeper, VA 22701-2780, johnny.chen@ariacoustics.com) and Jason E. Summers (Appl. Res. in Acoust., LLC, Washington, District of Columbia)

There is growing interest in prediction of anthropogenic noise levels in the ocean. Evidence suggests that sources of ambient noise such as shipping traffic may be destructive to marine organisms that rely on acoustics for communication. Characterizing and predicting ambient noise are also critical to effective naval operations. Understanding ocean noise is constrained by the limited ability to directly measure the spatial distribution of noise levels. In this work, we present a deep-learning method to estimate the spatial distribution of ambient noise due to shipping from a small number of measurements. Noise levels are typically estimated using forward models based on statistical information about shipping routes and source levels and predictions or measurements of environmental variables including bathymetry and sound-speed profile. Inverse methods are sometimes used to estimate input parameters from *in situ* data. In contrast, we demonstrate the robust estimation of shipping noise using a pretrained generative adversarial network (GAN) as a prior. By using a context and prior loss, our algorithm is able to accurately predict the entire spatial distribution of noise from sparsely sampled measurements. This can yield greater accuracy than a forward model based on environmental parameters taken from archival databases or inferred *in situ*.

3:00

1pAO8. Broadband acoustic propagation in a seagrass meadow throughout a diurnal cycle. Kevin M. Lee, Megan S. Ballard, Jason D. Sagers, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Gabriel R. Venegas, Jay R. Johnson, Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Abdullah F. Rahman (School of Earth, Environ., and Marine Sci., The Univ. of Texas Rio Grande Valley, Brownsville, TX)

Acoustic propagation in seagrass meadows is sensitive to gas produced by photosynthesis and respiration. In addition to gas volumes within the seagrass, bubbles are introduced into the water as oxygen diffuses through the plant tissue, leading to dispersion, absorption, and scattering of sound. Because the oxygen production cycle is largely driven by sunlight, these acoustical effects have a diurnal dependence. Previous work has examined the use of acoustics as a remote sensing tool for monitoring the seagrass photosynthetic activity (Hermand, 2004). In the present paper, we describe an acoustic propagation experiment conducted in a *Thalassia testudinum* meadow in the Lower Laguna Madre, a shallow bay on the Texas Gulf of Mexico coast. A spherical omnidirectional source transmitted frequency-modulated chirps (0.1 kHz to 100 kHz) every 10 min for a 24-h period, during which oceanographic probes measured water temperature, salinity, and dissolved oxygen. The received acoustic signals were match-filtered to obtain band-limited impulse responses, enabling identification of various propagation paths within the waveguide. The dependence of the received acoustic amplitude and frequency content on time-of-day, dissolved oxygen, and other environmental parameters will be discussed with the goal of using acoustics to study seagrass photosynthesis and productivity. [Work supported by ARL:UT IR&D and ONR.]

3:15

1pAO9. Effect of a soft sediment layer over elastic basement on normal mode dispersion. Gopu R. Potty, James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), and Julien Bonnel (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The effect of shear on dispersion of acoustic normal modes was investigated in a previous study (Potty and Miller, 2010). Modal dispersion was calculated using a bottom model consisting of a liquid layer over an elastic basement. The modal travel times corresponding to the Airy Phase regions were found to be extremely sensitive to shear. Simple inversion schemes were developed to estimate the shear speed in the sediment by comparing theoretical predictions with experimental data. Modal dispersion characteristics of broadband data collected during experiments conducted in Middle Atlantic Bight and New England Mud patch were analyzed, and bottom shear speeds were estimated. The estimated shear speeds were also compared with shear speeds calculated from core data. In this study, the bottom model will be revised to include a soft sediment layer over the elastic basement. The modal dispersion will be calculated using this bottom model corresponding to the New England Mud patch environment using propagation models such as ORCA and KRAKENC. The effect of the addition of the soft mud layer on dispersion of lower order modes will be investigated and presented. [Work supported by the Office of Naval Research.]

Session 1pBA

Biomedical Acoustics and Signal Processing in Acoustics: Lung Ultrasound and Tissue Stiffness Method II

Xiaoming Zhang, Cochair
Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

Libertario Demi, Cochair
Information Engineering and Computer Science, University of Trento, Via Sommarive, 9, Trento 38123, Italy

Chair's Introduction—1:20

Invited Papers

1:25

1pBA1. Machine learning assisted evaluation of interstitial lung diseases. Chi Wan Koo (Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, koo.chiwan@mayo.edu)

Interstitial lung disease (ILD) is an inflammatory condition encompassing greater than 200 different chronic disorders that often lead to pulmonary fibrosis and the attendant morbidity and mortality. Distinguishing one ILD from another can be challenging even for experts given similar clinical manifestations. However, firm diagnosis is important for management, counseling, and surveillance. Computer-Aided Lung Informatics for Pathology Evaluation and Ratings (CALIPER, Mayo Clinic, Rochester, MN, USA), a machine learned image analysis tool for characterizing and quantifying diffused lung diseases on CT, has been shown to correlate with ILD mortality such as for idiopathic pulmonary fibrosis. The CALIPER assessment of ILD is not influenced by confounding conditions such as emphysema or pulmonary hypertension that may affect pulmonary function measurements. Moreover, CALIPER can provide more reproducible results with less inter- and intraobserver variability compared to the conventional subjective visual assessment of CT scans by radiologists. Herein, we aim to provide an overview of ILD and discuss CALIPER evaluation of ILD.

1:45

1pBA2. Lung ultrasound surface wave elastography for assessing interstitial lung disease. Xiaoming Zhang, Boran Zhou, Jinling Zhou, Brian Bartholmai, Thomas Osborn, and Sanjay Kalra (Mayo Clinic , 200 1st St. SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

Many lung diseases including interstitial lung disease (ILD) are associated with changes in the lung's biomechanical properties. ILD comprises a number of serious diseases in which fibrosis stiffens and damages lung tissue. Most ILDs are typically distributed in the lung's peripheral and subpleural regions. Pulmonary function test (PFT) and high-resolution computed tomography (HRCT) are used to assess ILD. Ultrasonography is not widely used for lung assessment because ultrasound cannot image deep lung tissue. We have developed lung ultrasound surface wave elastography (LUSWE) for measuring superficial lung wave speed. In LUSWE, a 0.1-s harmonic vibration is generated on the chest wall of a subject using a handheld vibrator. An ultrasound probe is aligned with the vibration excitation in the same intercostal space to measure the generated surface wave propagation on the lung. A human subject is examined in a sitting position. The lung is tested at the total lung volume and through six intercostal spaces. Significant differences of surface wave speed between patients and controls were found in 6 lung regions and for 3 excitation frequencies. A positive correlation between LUSWE and clinical tests including HRCT and PTF was found. LUSWE may complement the clinical standard HRCT for assessing ILD.

2:05

1pBA3. Application of lung ultrasound surface wave elastography for assessment of extravascular lung water in patients hospitalized with congestive heart failure. Brandon M. Wiley (Cardiovascular Medicine, Mayo Clinic, 1216 Second St. SW, Rochester, MN 55905, wiley.brandon@mayo.edu), Boran Zhou (Radiology, Mayo Clinic, Rochester, MN), Govind Pandompatam (Cardiovascular Medicine, Mayo Clinic, Rochester, MN), Jinling Zhou (Radiology, Mayo Clinic, Rochester, MN), Hilal Olgun Kucuk (Cardiovascular Medicine, Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

Lung ultrasound (LUS) detects the presence of extravascular lung water (EVLW) through the visualization of B-Line artifacts. However, the qualitative nature of LUS limits its effectiveness in serial or longitudinal studies such as evaluating changes in EVLW at different time points in patients undergoing diuretic therapy for congestive heart failure. Lung ultrasound surface wave elastography (LUSWE) is a novel technique using a small handheld device that can measure superficial lung tissue elastic properties. We aimed to evaluate the use of LUSWE to measure quantitative changes in lung elasticity caused by acute changes in EVLW. We performed LUSWE on consecutive days in 14 patients hospitalized for acute congestive heart failure with evidence of pulmonary edema (clinical EVLW). From day#1 to day#2, the patients had an average diuresis of net negative 2.1 l associated with an average decrease in 13 B-

Lines by lung ultrasound, signifying a reduction in EVLW. LUSWE analysis demonstrated a significant reduction ($p < 0.05$) in surface wave velocity in all interrogated intercostal spaces from day#1 to day#2. In summary, LUSWE performed at the bedside was able to demonstrate improvement in lung compliance (decreased elasticity) correlating with a reduction in EVLW in hospitalized patients being treated for congestive heart failure.

2:25

1pBA4. Dedicated signal processing for lung ultrasound imaging: Can we see what we hear? Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive, 9, Trento 38123, Italy, libertario.demi@unitn.it)

The application of ultrasound imaging to the diagnosis and monitoring of the lung condition is nowadays receiving growing attention from both the clinical and technical world. The advantages of ultrasound are in fact numerous when compared to other imaging modalities such as CT: availability at patient site, low cost, real time, and safety. However, despite the vast amount of medical evidence showing how ultrasound can be used to gather diagnostic information, dedicated technical developments are still lacking. This leaves the clinitians with the only option of using standard ultrasound scanners and probes and consequently implies that decisions are often based on imaging artifacts. Standard ultrasound imaging is in fact based on the assumption that among the structures present in the field of view, only little variations in the speed-of sound are present. This is clearly not the case with the lung, due to the presence of air. In this talk, recently developed imaging modalities and signal processing techniques dedicated to the analysis of the lung response to ultrasound will be introduced and discussed. *In vitro* and clinical data will be presented which show how the study of the ultrasound spectral feature could lead to a quantitative ultrasound method dedicated to the lung.

Contributed Paper

2:45

1pBA5. Deep learning for automated detection of B-lines in lung ultrasonography. Ruud J. van Sloun (Eindhoven Univ. of Technol., Eindhoven, The Netherlands) and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive, 9, Trento 38123, Italy, libertario.demi@unitn.it)

The application of ultrasound imaging to the diagnosis of lung diseases is gaining attention. Of particular interest are several imaging-artifacts, e.g., A and B line artifacts. A-lines are hyperechoic horizontal lines, which are substantially visualized across the entire image and parallel to pleural-line. They represent the normal pattern of the lung if pneumothorax is excluded. Differently, B-line artifacts correlate with pathology and are defined as hyperechoic vertical artifacts, which originate from a point along the pleura-line and lie perpendicular to the latter. Their presence has been linked to an increase in extravascular lung water, interstitial lung diseases, non-cardiogenic lung edema, interstitial pneumonia, and lung contusion. In this work, we describe a method aimed to support the clinicians by automatically identifying the frames of an ultrasound video where B-lines are found. To this end, we employ modern deep learning strategies and train a fully convolutional neural network to perform this task on b-mode images of dedicated ultrasound phantoms (Demi *et al*., *Sci. Rep.* 2017). We moreover calculate neural attention maps that visualize which components in the image triggered the network, thereby offering simultaneous localization. Future work includes characterization of the detected B-lines to enable adequate phenotyping of various lung pathologies.

Invited Papers

3:00

1pBA6. Diagnosis and monitoring of pulmonary fibrosis using ultrasound multiple scattering, an *in vivo* rodent study. Kaustav Mohanty (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), John Blackwell, Mir H. Ali, Thomas Egan (Dept. of Cardiothoracic Surgery, Univ. of North Carolina, Chapel Hill, Chapel Hill, NC), and Marie M. Muller (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

Idiopathic pulmonary fibrosis (IPF) affects 200,000 patients in the U.S. IPF is responsible for changes in the micro-architecture of the parenchyma, such as thickening of the alveolar walls, which reduces compliance and elasticity. In this study, it is proposed to verify the hypothesis that changes in the micro-architecture of the lung parenchyma can be characterized by exploiting multiple scattering of the ultrasound waves by the lung parenchyma. Ultrasound propagation in a highly scattering regime follows a diffusion process, which can be characterized using the Diffusion Constant. We hypothesize that in a fibrotic lung, the thickening of the alveolar wall reduces the amount of air (compared to a healthy lung), thereby minimizing the scattering events. Pulmonary fibrosis is created in Sprague-Dawley rats by instilling bleomycin into the airway. The rats are studied in groups of $n = 6$ (3 male and 3 female) 2, 3, and 4 weeks after bleomycin administration. This allowed us to provide a range of severity of pulmonary fibrosis for assessment. Using a 128-element linear array transducer operating at 7.8 MHz, *in vivo* experimental data are obtained from Sprague-Dawley rats, and the Diffusion Constant is calculated. Right after the ultrasound measurement, the rats are euthanized, and computed tomography scans are performed to validate the degree of fibrosis created. Significant differences ($p < 0.05$) in the D values between control and fibrotic rats showcase the potential of this parameter for diagnosis and monitoring of IPF.

3:20–3:35 Break

3:35

1pBA7. Dynamic optical coherence elastography: Emerging tool for noninvasive quantification of mechanical properties of ocular tissues. Kirill Larin (Univ. of Houston, 4800 Calhoun Rd., 3605 Cullen Blvd., Rm. 2028, Houston, TX 77204, klarin@uh.edu)

Optical coherence elastography (OCE) is an emerging method for noninvasive quantification of tissue viscoelastic properties. The underlying technology is based on Optical Coherence Tomography (OCT) imaging and analysis of external (or internal) force-induced mechanical waves propagating through the tissue. In this presentation, I will overview recent progress made in my lab on quantification of mechanical properties of ocular tissues (such as cornea and the lens of the eye) using OCE and their alternations during diseases progression. In particular, I will demonstrate that it is possible to quantify mechanical properties of the cornea and lens as a function of Intra-Ocular Pressure (IOP) and during controlled tissue modifications and treatments, such as corneal cross-linking. These results indicate that OCE is a powerful new technology that can be utilized for the nondestructive biomechanical characterization of ocular tissues in normal and pathological states and could not only assist in basic biomechanical studies but also lead to a new class of optical sensors for diagnosis of diseases. [Work supported, in part, by the U.S. National Institutes of Health (NIH) under Grant Nos. 2R01EY022362 and 1R01HL120140 and U.S. Department of Defense (DOD) Congressionally Directed Medical Research Programs (CDMRP) under Grant No. PR150338.]

3:55

1pBA8. Comparison of ocular biomechanical properties in normal and glaucomatous eyes using ultrasound surface wave elastography. Arthur J. Sit, Arash Kazemi (Ophthalmology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, Sit.Arthur@mayo.edu), Boran Zhou, and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

Biomechanical properties of the eye are important in understanding glaucoma. However, specific tissues that may be affected are unclear. In this study, we compared glaucomatous and normal eyes for differences in corneal tissue elasticity (as indicated by wave speed) and global ocular rigidity. Both eyes of 10 glaucoma patients and 10 normal controls, matched for age and intraocular pressure (IOP), were included. The ocular rigidity coefficient was calculated from supine IOP measured with and without a 10 g weight added to the tonometer. The wave speed in the cornea was measured by ultrasound surface wave elastography. With this technique, a spherical-tipped probe (4 mm diameter) was placed on the closed eyelid and vibrated at 100 Hz for 0.1 s. The wave speed was calculated using the phase gradient method. Measurements for normal and glaucoma eyes were compared using generalized estimating equation models. The corneal wave speed was similar between normal and glaucomatous eyes ($P = 0.4$). However, ocular rigidity was significantly lower in glaucomatous eyes ($P < 0.001$) compared with normal eyes. There was no difference in age or IOP ($P > 0.3$). Lower ocular rigidity in glaucomatous eyes suggests that a more compliant ocular shell may predispose to glaucoma. However, the lack of difference in the corneal wave speed suggests that corneal tissue is not significantly affected, and changes likely involve the sclera.

4:15

1pBA9. Evaluation of posterior sclera viscoelasticity in eyes with papilledema by using ultrasound vibro-elastography. John J. Chen (Ophthalmology and Neurology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, chen.john@mayo.edu), Boran Zhou (Radiology, Mayo Clinic, Rochester, MN), Arash Kazemi, Arthur J. Sit (Ophthalmology, Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

Papilledema is optic nerve swelling caused by increased intracranial pressure, which has the potential to cause significant vision loss. Papilledema is typically bilateral and symmetric but can sometimes be asymmetric and even unilateral. The cause for this asymmetry is unknown. The purpose of this study was to utilize ultrasound vibro-elastography (UVE) to assess for biomechanical differences in eyes with papilledema. Nine patients with papilledema and 9 age-matched controls were enrolled. An external harmonic vibration was used to generate wave propagation through the eyelid with three excitation frequencies of 100, 150, and 200 Hz. A 6.4 MHz ultrasound probe was used to noninvasively measure the wave propagation in the posterior sclera to provide shear wave speeds and viscoelasticity (fit with Voigt model). The magnitudes of the shear wave speed and viscoelasticity of the idiopathic intracranial hypertension patients' posterior sclera were significantly higher than those of healthy subjects. Moreover, for patients with unilateral papilledema, the magnitudes of wave speed and viscoelasticity of the posterior sclera were statistically higher in eyes with papilledema than in the contralateral eyes without papilledema. UVE provides a noninvasive technique to measure the viscoelastic properties of the posterior sclera, which is stiffer in eyes with papilledema.

4:35

1pBA10. A comparison of hyperelastic constitutive models applicable to Shear Wave Elastography (SWE) data in tissue-mimicking materials. David Rosen and Jingfeng Jiang (Biomedical Eng. Dept., Michigan Technol. Univ., 1400 Townsend Dr., M&M 309, Houghton, MI 49931, jjiang1@mtu.edu)

Shear wave elastography (SWE) techniques have received substantial attention in recent years. Strong experimental data in SWE suggest that shear wave speed changes significantly due to the known acoustoelastic effect (AE). This presents both challenges and opportunities toward the *in vivo* characterization of biological soft tissues. In this work, under the framework of continuum mechanics, we model a tissue-mimicking material as a homogeneous, isotropic, incompressible, hyperelastic material. Our primary objective is to quantitatively and qualitatively compare experimentally measured acoustoelastic data with model-predicted outcomes using multiple strain energy functions. Our analysis indicated that the classic neo-Hookean and Mooney-Rivlin models are inadequate for modeling the AE in tissue-mimicking materials. However, a subclass of strain energy functions containing both high-order/exponential term(s) and second-order invariant dependence showed good agreement with experimental data. Based on data investigated, we also found that discrepancies may exist between parameters inversely estimated from uniaxial compression, and SWE data though Mooney plots were consistent between the uniaxial compression and AE results. Overall, our findings may improve our understanding of the clinical SWE results.

Contributed Paper

4:55

1pBA11. An ultrasound surface wave elastography technique for non-invasive measurement of scar tissue. Boran Zhou (Radiology, Mayo Clinic, 321 3rd Ave. SW, Rochester, MN 55902, Zhou.Boran@mayo.edu), Saranya P. Wyles, Alexander Meves (Dermatology, Mayo Clinic, Rochester, MN), Steven Moran (Plastic Surgery, Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

Hypertrophic scars and keloids are characterized by excessive fibrosis and can be functionally problematic. Indeed, hypertrophic scarring is characterized by wide, raised scars that remain within the original borders of injury and have a rapid growth phase. There is a need for quantitative scar measurement modalities to effectively evaluate and monitor treatments. We aim to assess the role of the non-invasive scar-measuring device, ultrasound surface wave elastography (USWE), in accurately evaluating scar metrics. Two sites of breast tissue were tested control and scar portions. In USWE, a small, local, and 0.1-s harmonic vibration at three excitation frequencies (100, 150, and 200 Hz) was generated on these sites, and the resulting surface wave speed was measured via an ultrasound probe with a central frequency of 6.4 MHz. There was a statistically significant difference in the wave speed at three frequencies of the scar portion between prior and after treatment, suggesting that the scar portion was softener after treatment. USWE provides an objective assessment of the reaction of the scar to injury and treatment response.

MONDAY AFTERNOON, 13 MAY 2019 BREATHITT, 2:00 P.M. TO 4:35 P.M.

Session 1pMU

Musical Acoustics and Signal Processing in Acoustics: Transient Phenomena in Wind Instruments

Vasileios Chatziioannou, Chair
Department of Music Acoustics, University of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Building M, Vienna 1030, Austria

Chair's Introduction—2:00

Invited Papers

2:05

1pMU1. Multiphonic modeling using impulse pattern formulation. Simon Linke, Rolf Bader (Systematic Musicology, Univ. of Hamburg, Finkenau 35, Hamburg 22081, Germany, simon.linke@haw-hamburg.de), and Robert Mores (Media Technic, Hamburg Univ. of Appl. Sci., Hamburg, Germany)

Multiphonics, the presence of multiple pitches within the sound of wind-instruments can be produced in several ways. Either complex fingerings are used or the blowing pressure is very low or very high. Such multiphonics can be modeled by the Impulse Pattern Formulation (IPF) proposed previously [R. Bader, *Nonlinearities and Synchronization in Musical Acoustics and Music Psychology* (2013)]. This top-down method assumes musical instruments to work with impulses which are produced at a generator, travel through the instrument, are reflected at various positions, are exponentially damped, and finally trigger or at least interact with succeeding impulses produced by the generator. While modeling sounds produced at blowing-threshold, the IPF fully captures transitions between regular periodicity at nominal pitch, bifurcation scenarios, and noise, just like regular instruments do, when multiphonics appear in the transition regime. Using IPF, complex fingerings translate to multiple reflection points at open finger holes with different reflection strengths. Here, complex multiphonics can be modeled. The IPF can also synthesize multiphonic sounds when applying the typical impulse form of wind instruments.

2:25

1pMU2. Initial transients in free reed instruments: A survey of experimental results. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

For free reeds in steady oscillation, the fundamental transverse beam mode dominates, but transverse modes and some torsional modes are also typically present. The motion of the free-reed tongue in early stages of the initial transient has been studied experimentally with the aim of determining the presence of higher modes and their possible role in the attack transient. These transients have been studied for free reeds mounted on a wind chamber with several methods used to initiate the attack transients. The resulting displacement and velocity waveforms have been studied using laser vibrometry, variable impedance transducer proximity sensors, and high speed video with the tracking software. The most realistic procedure used a pallet valve mechanism simulating the attack transient in a key-

operated instrument. Short-term spectra derived from the waveforms have been analyzed, showing that both higher transverse modes and some torsional modes are observed in the initial transient, with the second transverse mode and the torsional mode especially prominent in the earlier stages of oscillation. Comparisons of reed tongues of different designs have been made to explore the role of these modes in the initial excitation.

2:45

1pMU3. Detecting articulations in clarinet playing. Jack D. Gabriel and Whitney L. Coyle (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, jgabriel@rollins.edu)

A method is suggested to automatically distinguish slurred transitions from tongued transitions in clarinet playing based on the mouth pressure signal. Player data will be presented from musicians playing a clarinet with a sensor-equipped mouthpiece. Playing parameters such as blowing pressure and pressure in the mouthpiece were captured, and data recorded in controlled tests and musical passages will be presented. Possible future applications of the method will be discussed, such as automatic detection of the quality clarinet and clarinet accessories.

3:05

1pMU4. Reproducing tonguing strategies in single-reed woodwinds using an artificial blowing machine. Montserrat Pàmies-Vilà, Alex Hofmann, and Vasileios Chatziioannou (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria, pamies-vila@mdw.ac.at)

Articulation on woodwind instruments is achieved inside the player's mouth, where the tongue interacts with the vibrating reed, while the player adjusts the blowing pressure, the lip force, and the vocal tract configuration. The performed articulation technique defines the characteristics of the attack and release transients and thus the transitions between tones. In this study, an artificial blowing machine with a built-in tonguing system is used to analyze different tonguing strategies in single-reed woodwinds. The tonguing system is controlled via an electronically monitored shaker, offering the possibility to reproduce tongue articulation, while assuring repeatability. To reproduce different playing techniques, parameters obtained from measurements with players are used to set up the pressure in the artificial mouth and the behaviour of the tonguing system. During the experiment, the artificial-mouth pressure, the mouthpiece pressure, the reed displacement, and the shaker acceleration are recorded. The recorded signals are then compared to real-playing clarinet and saxophone measurements. Different trajectories for the artificial tongue are tested as well as different tongue-reed-contact durations. The artificial blowing and tonguing set-up, along with the images obtained with a high-speed camera, provide an in-depth understanding of the processes taking place inside the player's mouth.

3:25–3:40 Break

3:40

1pMU5. Imaging of transient and steady-state flow in organ pipes. Thomas Moore and Whitney L. Coyle (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

The transient and steady-state flow at the flue and open end of organ pipes has been studied using high-speed electronic speckle pattern interferometry. We show that the flow at the flue is similar to square and round organ pipes; however, at the open end, the flow depends on the pipe shape. There is a significant flow from a square pipe and no detectable flow from a round pipe. These results can be directly compared to the results of simulations recently developed by Thacker and Giordano. [Work supported by NSF Grant #PHY-160749.]

4:00

1pMU6. Transients in wind instruments modeled with the Navier-Stokes equations. Nicholas Giordano (Phys., College of Sci. and Mathematics, Auburn Univ., Auburn, AL 36849, njg0003@auburn.edu)

Transient components make an important contribution to the "color" of a musical tone. While such transients can be observed in experiments, realistic modeling can be very challenging. We describe a modeling study of initial transients in the tone produced by a recorder using Navier-Stokes-based simulations. We have studied how the harmonic content of a recorder tone during the attack portion of the tone depends on a variety of factors including the labium position, the presence of chamfers at the exit of the windway, and the initial dynamics of the blowing pressure. While our results are obtained for a realistic model of the recorder, they should also be applicable to similar instruments such as flue organ pipes. [Work supported by NSF Grant PHY1513273.]

Contributed Paper

4:20

1pMU7. Computational studies of flow in flue pipes. Jared W. Thacker and Nicholas Giordano (Phys., Auburn Univ., 210 East Thach Ave., Apt. 24E, Auburn, AL 36830, jwt0024@tigermail.auburn.edu)

Direct numerical solutions of the Navier-Stokes (NS) equations have been used to study the air flow in a recorder. When the recorder is driven with a DC flow into the windway, we observe the familiar oscillating flow out of the window adjacent to the labium. The associated flow at the open end is also studied. For a pipe with a square cross-section, we find the flow at the open end to be small but nonzero, consistent with recent studies using speckle pattern interferometry by Coyle and Moore. In addition, we have analyzed the NS results for the air density so as to compare directly with the interferometry experiments. We also describe simulations for circular pipes. [Work supported by NSF Grant PHY1513273.]

Session 1pNS

Noise, Psychological and Physiological Acoustics: Acoustic Vehicle Alerts: Effects on Soundscape, Quality of Life, and Traffic Safety

Jeanine Botta, Cochair
SUNY Downstate Medical Center School of Public Health, 720 East 31st Street, Apartment 7G, Brooklyn, NY 11210

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

Chair's Introduction—1:30

Invited Papers

1:35

1pNS1. Acoustic vehicle alerts and sleep disruption: A content analysis of online complaints and inquiries. Jeanine Botta (The Right to Quiet Society for Soundscape Awareness and Protection, 720 East 31st St., Apartment 7G, Brooklyn, NY 11210, jeanine.botta@downstate.edu)

Research on the effects of traffic noise on sleep focuses on sounds created by vehicles that are traveling, whether in continuous motion on a highway or road or temporarily slowed on a congested street. Most acoustic vehicle alerts are introduced without consideration for their potential to disrupt sleep or affect quality of life. Panic alarm, marketed as a safety feature and standard with most cars, was accepted by regulatory and consumer protection agencies without question or concern for its necessity or its potential to create new noise. Criticism about car alarms, remote lock signals that use horn sounds, panic alarm, and backup beeping in passenger cars is common in everyday discourse but rare within the public health sphere. This study will use content analysis of online complaints posted in public databases, discussion forums, and car owner forums to explore common experiences related to sleep disruption. Focus will be on technologies not used while driving and will not include pedestrian alerts or backup beeping used by commercial or industrial vehicles. The study will include posts by those whose sleep has been affected and by car owners inquiring about methods of turning off a sound source out of concern for a neighbor.

1:55

1pNS2. Soundscape, traffic safety, and requirements for public health. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

In October 2018, the World Health Organization has published the Noise Guideline for the European Region strongly focusing on health effects caused by noise from different sources as transportation (road traffic, railway, and aircraft) noise, wind turbine noise, and leisure noise. As outlined in the Introduction, they "provide robust public health advice underpinned by evidence, which is essential to drive policy action that will protect communities from the adverse effects of noise." Otherwise, the new technology in the development of electrical vehicles causes regulations calling for safety reasons for alert signals that may be counterproductive with regard to a harmonic and healthy soundscape. Regulations and needs will be discussed with respect to the public health recommendations on exposure to environmental noise and soundscapes.

2:15

1pNS3. The conflict of soundscape requirements with the introduction of alert signals. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

Quiet road vehicles have to be equipped with acoustic alert signals for type approval to account for their potentially reduced audibility. According to the FMVSS No. 141, the alert sound has to be recognizable as a motor vehicle in operation that allows blind and other pedestrians to detect nearby electric vehicles or hybrid vehicles operating at lower speeds. Unfortunately, the detailed minimum sound requirements for type approval vary from regulation to regulation, for example, between FMVSS No. 141 and UNECE No. 138. The manufacturers can consider brand-related alert signal design within certain ranges but must adapt their alert sounds to the respective regulations, because an international harmonization of the signal requirements is missing. In this context, the impact on traffic noise in general and on soundscape in particular seems to be rarely systematically discussed. It is expected that some side issues will emerge with the systematic introduction of alert signals. Moreover, the effectiveness of the different regulations regarding the significant reduction of pedestrian collision risk was not systematically analyzed so far. This paper will discuss potential shortcomings of the introduction of alert signals regulated by law, and the implications on urban soundscapes will be outlined.

2:35

1pNS4. How to design vehicle alert signals? Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

E-Mobility becomes more and more popular. At the same time, acoustical alert signals are requested by law. This is in conflict with less traffic noise and better environmental sound. Especially, the request that the alert signals should contain tones, and these tones should have a frequency shift in the dependency of speed can create disharmonic sounds, e.g., if more than one vehicle is present. This will be a negative impact on the soundscape, and the annoyance will increase. Different concepts of alert signals will be discussed with respect to detectability, localization, and sound quality.

2:55

1pNS5. Should hybrid and electric vehicles have acoustic alerting systems? Rene Weinandy (Noise Abatement in Transport, German Environment Agency, Woerlitzer Platz 1, Dessau-Rosslau 06844, Germany, rene.weinandy@uba.de), Lars Schade, and Jan Gebhardt (Noise Abatement in Transport, German Environment Agency, Dessau-Rosslau, Saxony-Anhalt, Germany)

Noise is an oft-overlooked environmental issue within densely populated regions. Vehicles, railways, and airports operating within or near cities are all contributing to the growing noise pollution problem—causing negative health and economic impacts. Due to this, it is of primary importance to make our cities quieter. The German Environment Agency is working on noise and its effects on humans, especially with respect to policy in order to make traffic as quiet as possible by addressing all the relevant elements from roads and tracks to vehicles, operational procedures, and measures along the sound propagation path. Europe, as well as most of the world, faces a future full of environmentally friendly hybrid or pure electric road vehicles. Concerns were raised that these low-emission vehicles could pose a risk to blind and low vision pedestrians. To address this concern, the European Union has legislated that future hybrid and pure electric cars must be equipped with acoustic vehicle alerting systems (AVAS). The presentation provides a critical assessment of the effectiveness of AVAS and of their negative side effects. Furthermore, it explores alternative non-acoustic approaches addressing aspects such as environmental protection, road safety, feasibility, and usability.

3:15

1pNS6. Upgrading backup alarms to reduce encroachment on soundscapes in Denali National Park. Davyd Betchkal (Natural Sounds and Night Skies Div., National Park Service, MP 237 Parks Hwy., P.O. Box 9, Denali Park, AK 99755, davyd_betchkal@nps.gov) and Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

Denali maintains a sizeable fleet of maintenance vehicles that are equipped with backup alarms. Across many years, park acoustical monitoring demonstrated that tonal backup alarms were audible at one to two miles distance from the hazard zone immediately behind the truck. Following a successful demonstration of backup alarms with broadband signals, 77 vehicles at Denali were upgraded. The aggregate cost of the devices was just under $11,000, and the Denali auto shop provided 25 h of staff time for the upgrade. CadnaA modeling was used to compare the park area in which the old and new alarms were audible, assuming that both broadcast the same level (107 dB, A-weighted). The broadband alarm was predicted to be audible in 2.9 km^2, compared with 6.9 km^2 for the tonal alarm. The adaptive level feature of the new alarms yielded additional benefits in quiet locales. After the upgrade, backup alarm noise became noticeably less along the Denali Road corridor, and these improvements motivated the park to incorporate language regarding backup alarms into construction contracts.

Session 1pPAa

Physical Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, Noise, Psychological and Physiological Acoustics, and Speech Communication: Battlefield Acoustics II

W. C. Kirkpatrick Alberts, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20723

Gregory W. Lyons, Cochair
Construction Engineering Research Laboratory, U.S. Army Engineer Research and Development Center, 2902 Newmark Dr., Champaign, IL 61822

Chair's Introduction—1:00

Invited Papers

1:05

1pPAa1. Hearing on the battlefield: The challenge of protecting hearing while enabling the mission. Eric R. Thompson and Brian Simpson (711th Human Performance Wing, Air Force Res. Lab., 2610 Seventh St., B441, Wright-Patterson AFB, OH 45433, eric.thompson.28@us.af.mil)

The auditory sense plays a critical role in human performance on the battlefield. Effective verbal communication is key to the success of all military operations, the ability to detect and localize sounds in the immediate environment is critical for supporting situation awareness, and the ability to identify sound sources is required for labeling items and events as hostile or friendly. However, operational environments tend to be extremely noisy, and most warfighters are regularly exposed to hazardous noise levels, necessitating the use of hearing protection to reduce the chance of experiencing temporary or permanent hearing loss. Unfortunately, the use of hearing protection can have a deleterious effect on communication, sound source identification, and overall situation awareness. For this reason, operators often choose to perform all, or part, of their mission without the appropriate hearing protection in place. In order to protect warfighters' hearing, appropriate hearing protection needs to be selected that will provide enough protection from the expected hazardous sounds, while also enabling the hearing-critical tasks that are required for the mission. This presents a great challenge for those who make the decisions on what hearing protection to prescribe.

1:25

1pPAa2. Improving speech intelligibility performance for aircraft maintainers by selecting an appropriate combination of hearing protection devices. Hilary Gallagher (Battlespace Acoust. Branch, Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, hilary.gallagher.1@us.af.mil) and Billy Swayne (Systems Eng. Solutions, Ball Aerosp. and Technologies Corp., Wright-Patterson AFB, OH)

Aircraft maintainers work in hazardous noise environments throughout their career. Hazardous noise levels on the flight line vary, affecting the level of hearing protection required to reduce their risk of hearing loss and other hearing related disabilities. Current practice is to provide double hearing protection: a circumaural headset/earmuff with foam earplugs. This combination may provide an adequate amount of protection from aircraft noise, but the proper use of foam earplugs may degrade speech intelligibility (SI) performance. Due to this degradation, many maintainers fit the foam earplugs incorrectly in order to improve SI performance, therefore lowering the amount of protection provided. The objective of this study was to assess the noise attenuation and SI performance of a headset and filtered earplugs worn alone and in combination to provide a more appropriate recommendation of hearing protection devices. The filtered earplug can provide suitable levels of protection when worn alone or in combination with a headset (depending on the noise environment) and can provide adequate "hear through" capability for acceptable SI performance when communicating face-to-face (single hearing protection) or through the aircraft communication system (double hearing protection).

1:45

1pPAa3. Acoustic validation of military aircraft noise models. Alan T. Wall and Frank S. Mobley (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com)

The National Environmental Policy Act (NEPA) of 1970 established the requirement to assess noise impacts from aircraft operations on the community surrounding military bases. A process for measuring and modeling aircraft noise sources and their propagation to ground locations was developed in response. The NOISEMAP suite of software programs produces noise footprints of yearly average impacts due to all flights that occur to and from an airbase. Through NOISEMAP's decades of evolution, it retains the same fundamental measurement/modeling architecture and remains the approved model for environmental impact studies for aircraft within the U.S.

Department of Defense (DoD). However, new advanced source and propagation models exist, which result in lower uncertainties and the calculation of additional useful noise impact metrics. No clear path exists to allow the use of these models for DoD environmental impact studies. This work presents a scientific validation experiment for aircraft overflight source/propagation models, focusing on suitability for use in a NEPA environmental impact study. The experiment relies on high-fidelity data collected during controlled overflight acoustic measurements of real aircraft and provides a quantitative benchmark for the evaluation of future models.

2:05

1pPAa4. Acoustical modeling considerations in noise assessment of live fire training ranges. Michael J. White and Michelle E. Swearingen (US Army ERDC/CERL, P.O. Box 9005, Champaign, IL 61826, michael.j.white@usace.army.mil)

In the US, long-term noise assessments near military firing ranges are required to estimate the weighted average of simulated sounds from thousands of noise-producing activities. Each of these activities are modeled with a set of parameters and heuristics. Parameters include source type, including quantification of firing and munition explosive charges; location information; high or low angle of fire designation; and basic environmental conditions including time of day, time of year, generalized surface type, and some meteorological information. Heuristics model the source strength and directivity, divergence of waves, ground reflection, atmospheric refraction, absorption of sound, and shielding by noise barriers and terrain. The heuristics represent limitations to the fidelity of the modeling result, and they produce estimates with significant uncertainties. In view of the uncertainties, validation of results is particularly difficult and must be performed on single activity calculations rather than long-term averages. To evaluate the uncertainty, we compare single event calculations to controlled measurements, thereby minimizing the parameter uncertainty. These ideas are discussed, and an assessment of the model uncertainty is presented.

Contributed Papers

2:25

1pPAa5. Nonlinear tuning curve demonstration: Comparing a buried mine simulant with a clamped elastic plate—Cylindrical soil column oscillator. Ava B. Twitty (Phys. Dept., U.S. Naval Acad., 5568 Carvel St., Churchton, MD 20733, avat1226@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

Experiments using soil-plate-oscillators (SPO) involve a cylindrical column of granular media (masonry sand) supported by a clamped circular elastic acrylic plate (12.7 cm diameter, 3.2 mm thick). The plate is clamped between two 20.3 cm O.D., 12.7 cm I.D., 6.4 cm thick flat toroidal brass "rings." Two 15 cm diameter subwoofers (10 cm above the soil) are driven by an amplified swept sinusoidal chirp which drives the 2.5-cm soil column. A spectrum analyzer measures the laser Doppler vibrometer particle velocity versus frequency near the center of the column. The resonant frequency decreases with the increasing amplitude—representing softening in the nonlinear system. The back-bone curve (locus of the resonant frequency versus corresponding peak velocity coordinates) has a distinct arching shape where the slope of the velocity increases with the decreasing frequency. Here, the resonant frequency goes from 247 to 203 Hz. A lumped element bilinear hysteresis model describes the shape of the tuning curves and backbone curve. Next, a drum-like simulant (made by replacing the upper toroidal ring by a 0.64-cm thick ring) is buried 2.5-cm deep in an open square concrete tank (57 cm). Nonlinear tuning curve experiments "on" and "off" the mine are compared with the SPO results.

2:40

1pPAa6. Inter-array coherence-based identification of static acoustic clutter sources. W. C. Kirkpatrick Alberts (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20723, william.c.alberts4.civ@mail.mil)

In multi-array acoustic detection and tracking applications, there often exist valid acoustic sources that are unwanted because of their position or nature, e.g., generators and air-handling equipment. During a tracking scenario where desired targets can be moving and/or piecewise static, the presence of these clutter sources can negatively impact the localization of a desired target by interrupting a track or generally increasing the noise floor local to an array. In directions away from the clutter source, beamforming methods onboard an array, e.g., minimum variance distortionless response, can be used to suppress the signature of the unwanted source. However, when a desired target passes through the azimuth of a clutter source, beamformers may fail to separate the desired source from the clutter. Here, inter-array coherence between beamformed solutions will be used to identify a static source by its properties in order to remove the clutter source as a potential target in a tracking application. Acoustic data on clutter sources measured by multiple microphone arrays separated by tens of meters will be discussed.

Session 1pPAb

Physical Acoustics, Archives and History, and Education in Acoustics: On His 100th Birthday, Isadore Rudnick Speaks for Himself

Julian D. Maynard, Chair
Physics, Penn State University, 104 Davey Lab, Box 231, University Park, PA 16802

Chair's Introduction—3:15

Invited Papers

3:20

1pPAb1. Isadore Rudnick's spectacular acoustics demonstrations. Robert M. Keolian (Sonic Joule LLC, 732 Holmes St., State College, PA 16803, keolian@psu.edu), Steven R. Baker (Phys., Naval Postgrad. School, Marina, CA), and Arthur Huffman (Phys., Univ. of California, Los Angeles, CA)

While soft-spoken in conversations and lectures, Isadore Rudnick could shout when teaching acoustics with demonstrations. At a special, auditorium stage, video-taped plenary session of the Fall 1980 meeting of the Acoustical Society of America, Izzy presented 90 min of spectacular acoustic demonstrations. The demonstrations ranged from simple ones common to acoustics classes to ones that filled the stage and involved elaborate equipment. This talk will be followed by a viewing of Izzy's plenary session video. It is hard to imagine that any acoustician would not find something of interest in, and indeed learn something from, this video. For those who wish to study Izzy's demonstrations further or present the material in a classroom, the video is available for ordering as advertised in the Journal of the Acoustical Society of America.

5:00

1pPAb2. The unusual properties of liquid helium. Steven L. Garrett (151 SycamoreDr., State College, PA 16801, sxg185@psu.edu)

Each year since 1925, one UCLA faculty member has been selected to present the Faculty Research Lecture. In 1976, Isadore Rudnick was so honored. As an experimentalist, Izzy chose to make his presentation a lecture demonstration, like those of Michael Faraday's famous Christmas presentations at the Royal Society in London. This required that an entire low-temperature laboratory be recreated on the stage of Schoenberg Hall, including a transparent Dewar vessel. The success of those live demonstrations, and the fact that two of Izzy's sons were film makers, prompted him to make a 17-min film version of those demonstrations. That film won the award for best Technical and Scientific Film at the 21st Annual San Francisco International Film Festival, in 1977. Also, at this same time, the Cultural Revolution in China had ended, and Izzy was one of the first U.S. physicists invited to lecture there. As a gift to his Chinese hosts, he had this film translated into Mandarin. Due to the scarcity of the material available for those early Chinese television broadcasts, the film was played frequently, and Izzy became a TV celebrity throughout China. The film, shown in this talk, is included in ASA's *Collected Works of Isadore Rudnick*.

Session 1pPP

Psychological and Physiological Acoustics, Speech Communication, and Signal Processing in Acoustics: Applications of Signal Detection Theory in Perception and Physiology

Jennifer Lentz, Cochair
Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405

Christopher Conroy, Cochair
Boston University, 635 Commonwealth Ave., Boston, MA 02215

Contributed Papers

1:05

1pPP1. Lessons learned from signal-detection-theory-oriented studies of the processing of auditory patterns. Charles S. Watson (Speech and Hearing Sci., Indiana Univ., CDT, Inc., 3100 John Hinkle Pl, Bloomington, IN 47408, watson@indiana.edu)

One extension of Fechner's (1860) methods by which the influence of sensory stimuli on human observers could be quantified in physical units, sometimes termed a new psychophysics was Stevens (1938) theory and methodology of perceived magnitude estimation that attempted to bridge the gap between sensation and perception. A more far-reaching second extension began in 1954, when Peterson, Birdsall, and Fox applied statistical decision theory to the recovery of physical signals embedded in background noise and Tanner, Green, and Swets applied those principles to develop a new psychophysical methodology. What became known as "signal detection theory" provided methods of determining the capacity of systems to detect, discriminate, or recognize physical stimuli, a second new psychophysics. The two alternatives to Fechner represent complimentary approaches to a single set of phenomena, one of which can be characterized as the study of *sensory capabilities* and the other of *response proclivities*. The two require entierly different methods of measurement and address fundamentally different questions. Both are of practical and theoretical importance. Levels of stimulus uncertainty and the type and duration of experience with stimuli can have profound influences on measurements of both types. Examples of these effects for complex sounds, both speech and nonspeech, will be presented.

1:20

1pPP2. Signal detection theory and psychoacoustics. Christopher Conroy (Dept. of Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cwconroy@bu.edu) and Jennifer Lentz (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

This talk will trace the early history of signal detection theory (SDT) with a particular emphasis on its applications in psychoacoustics. SDT, as developed at the University of Michigan (U-M) in the mid-1950s, revolutionized psychophysics by introducing a core assumption: all sensory judgements are limited by "noise" of one sort or another and, as such, are reflective of underlying decision processes. To test and flesh-out this new theory, early empirical work in vision quickly turned to audition and a remarkably fruitful 12-year period followed, from roughly 1954 to 1966. This talk will focus on that 12-year period, bracketed, on the one end, by the establishment of an auditory research laboratory at U-M by Wilson P. "Spike" Tanner and, on the other, by the publication of Green and Swets' (1966) seminal text. The principle documents and publications of that period—particularly those that appeared in this society's *Journal*—will be reviewed, with a discussion of their intellectual and historical context. Emphasis will be given to the investigations of the Electronic Defense Group at U-M and those of other legendary psychoacousticians. In particular, the work of Tanner, the key instigator in the transition from "statistical decision theory" to SDT in psychoacoustics, will be discussed.

Invited Papers

1:35

1pPP3. A detection-theoretic approach to cocktail-party listening. Robert Lutfi, Brianna Rodriguez (Dept. Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL 33620, rlutfi@usf.edu), and Jungmee Lee (Commun. Sci. and Disord., Univ. of South Florida, Madison, WI)

Cocktail-party listening (CPL) is a key-term referring to difficult listening situations wherein one must h" and follow the speech of individual talkers in a crowd. Such situations are commonplace in everyday listening but can be a challenge to study. Important factors affecting performance, such as voice similarity among talkers and uncertainty associated with the dynamic variation of speech, can be difficult to quantify. Spectral and spatial acoustic cues distinguishing talkers are of critical importance but are expressed in different physical units making their relative role difficult to evaluate beyond the particulars of a study. There are also often huge individual differences in listener performance that can complicate the interpretation of results. This paper reviews recent applications of detection theory designed to address these challenges. The distinguishing feature of the approach is that key factors are evaluated in terms of their contribution to the information divergence of talkers, D_{KL}; a single statistic that dictates optimal performance for each task [Lutfi *et al*., *J. Acoust. Soc. Am.* **134**, 2160–2170 (2013)]. Both published and unpublished studies are reviewed demonstrating the application of the approach to each of the major challenges described above. [Work supported by NIDCD R01-DC001262].

1:55

1pPP4. On the theoretical benefit of combining multiple observations: Dependence on the form of the probability distribution of the observations. Huanping Dai (Speech Lang. and Hearing Sci., Univ. of Arizona, P.O. Box 21071, 1131 E. 22nd St., Tucson, AZ 85721-0071, hdai@email.arizona.edu) and Emily Buss (Dept. of Otolarynology/Head and Neck Surgery, Univ. of North Carolina, Chapel Hill, Chapel HIll, NC)

Auditory research has benefited immensely from the application of the Signal-Detection Theory (SDT; Green and Swets, 1966). The theory has provided a rigorous framework within which the design of experimental studies and the interpretation of experimental results can be carried out in a principled way. A fundamental assumption often made, either explicitly or implicitly, in the application of SDT is that the observations obey Gaussian distributions. Many important results were derived on an equal-variance Gaussian assumption for the "noise" and the "signal"events. One example is the well-known relation between the detectability attainable based on a single observation (d'_1) and the detectability that can be achieved when multiple ($m > 1$) independent and equally informative observations are combined: $d'_m = \sqrt{m}\, d'_1$. As researchers often need to deal with situations where the form of the probability distributions of the observations either is unknown or is known to be non-Gaussian, it is important to understand how the form of the probability distribution can influence the outcome. This presentation will examine different ways by which the interpretation of listeners' performance with multiple observations would depend on the form of the probability distributions.

2:15

1pPP5. Physiological parallels with perception. John C. Middlebrooks (Otolaryngol., Univ. of California at Irvine, Rm. 116 Medical Sci. E, Irvine, CA 92697-5310, j.midd@uci.edu)

Signal detection theory (SDT) lends itself to comparison among perceptual and physiological results in humans and other animals. In physiological recordings, trial-by-trial distributions of single-unit spike counts or of event-related-potentials typically yield non-normal distributions, calling for non-parametric tests. An empirical receiver operating characteristic (ROC) curve can compare distributions of physiological measures between pairs of stimulus conditions. The area under the ROC curve is a non-parametric measure of sensitivity, yielding the proportion correct and, thence, the sensitivity index, *d'*. The *d'* cumulated over a succession of the increasing values of a stimulus parameter provides a measure of the growth of response over that parameter range. I will present examples from several ongoing projects including: (1) studies of activation of the central auditory pathway by cochlear implants and by a penetrating auditory nerve electrode, with single-unit recordings from the inferior colliculus; (2) comparisons of spatial stream segregation among 2-alternative forced choice in humans, hold-response measures in cats, and cortical single-unit recording in cats; and (3) studies of frequency sensitivity using the auditory change complex and masked onset responses. In general, these techniques derived from SDT yield a satisfying agreement among perceptual measures and their physiological underpinnings.

2:35

1pPP6. Accounting for a wide variety of binaural detection data by combining cross-correlation and signal-detection theory approaches. Leslie R. Bernstein and Constantine Trahiotis (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, Farmington, CT 06032, lbernstein@uchc.edu)

We will discuss several experimental contexts in which a Theory of Signal Detection (TSD) approach to modeling has yielded successful, straightforward, and intuitively appealing accounts of data concerning binaural auditory processing. The primary focus will be on recently published empirical data and quantitative modeling from our laboratory. Those reports demonstrate that data obtained in binaural detection experiments conducted across the last five decades can be accounted for by combining a signal-detection-based decision variable with a cross-correlation-based model of binaural processing that incorporates stages of peripheral auditory processing. Notably, some of the data obtained in those experiments had, therefore, been "problematic" in that they had remained either theoretically unaccounted for or, at best, had been only accounted for via *ad hoc* approaches. Key to developing the unified account of those experimental results was the calculation and inclusion of the variability of the interaural correlations of the outputs of the model for both masker-alone and signal-plus-masker conditions. We will also discuss recent developments showing how our TSD-inspired approach has proven useful in discovering and explaining why some listeners with slightly elevated, but clinically negligible, audiometric thresholds exhibit reliable and meaningful deficits in binaural performance. [Work supported by Office of Naval Research (N00014-15-1-2140 and N00014-18-1-2473).]

2:55–3:10 Break

3:10

1pPP7. Coping with the black swan in psychophysics. Yi Shen (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S. Jordan Ave., Bloomington, IN 47405, shen2@indiana.edu)

Aberrant responses that are associated with lapses in attention and microsleeps during a behavioral experiment violate basic assumptions underlying signal detection theory. These responses (i.e., the black swan events) occur infrequently but could lead to severe biases in the estimated perceptual sensitivity (d') and misinterpretations of the data. When adaptive procedures are used to estimate behavioral thresholds, performance lapses occurring early in an adaptive track may cause convergence failures. Moreover, microsleeps may become more frequent as the task difficulty decreases, leading to potential instabilities in common adaptive procedures. Through a series of Monte-Carlo simulations, the potential effects of performance lapses on typical psychophysical tasks (e.g., Yes/No, 2-alternative forced choice) were demonstrated. Several possible experimental and computational techniques to cope with performance lapses were investigated. These included (1) introducing additional parameters representing performance lapses into the psychometric model, (2) allowing a "Not Sure" response category, and (3) screening and removing the likely aberrant responses. These simulation studies provide useful guidelines to design adaptive psychophysical procedures for subject populations from which performance lapses are expected (e.g., young children and behaving animals).

3:30

1pPP8. Extending signal detection theory approaches up the central pathway: Auditory midbrain models. Laurel H. Carney (Biomedical Eng. & Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu) and Miriam Furst (Elec. Eng., Tel Aviv Univ., Tel Aviv, Israel)

The application of signal detection theory (SDT) to peripheral encoding of stimulus parameters provided a quantitative basis for understanding auditory-nerve (AN) responses. Siebert's development of this strategy, based on relatively simple analytical models, laid the groundwork for Heinz's extension that took advantage of computational AN models. These studies applied the Cramer-Rao lower bound (CRLB) to estimate just-noticeable differences of stimulus parameters, assuming that underlying neural responses could be described as non-homogenous Poisson processes (NHHP) and that the brain acts as an optimal processor. Krips and Furst carried this approach into the central nervous system, first showing that coincidence-detector neurons that receive excitatory and/or inhibitory NHHP inputs produce outputs that are also NHHP. These models were tested using the CRLB for sensitivity to interaural differences in time and level. Here, we carry the approach a step further along the auditory pathway to the auditory midbrain. Model neurons that combine excitatory-excitatory and etxcitatory-inhibitory mechanisms explain rate tuning for the amplitude-modulation frequency, a key response property of inferior colliculus (IC) neurons. These models also exhibit direction selectivity for fast frequency sweeps observed in IC responses to Schroeder-phase harmonic complexes. These models facilitate SDT predictions of psychophysical tasks involving amplitude and frequency fluctuations.

3:50

1pPP9. Data-driven methods in a model-driven world: Making sense of auditory classification images for continuous speech. Jonathan Venezia (Auditory Res., VA Loma Linda Healthcare System, 11201 Benton St., Loma Linda, CA 92357, jonathan.venezia@va.gov)

Some of my recent work has focused on the development of Auditory Bubbles, a method for generating classification images that show the acoustic features underlying different percepts of continuous speech. Auditory Bubbles work by applying randomly shaped spectrotemporal filters to the modulation power spectrum of sentence-level speech stimuli. A reverse correlation is then used to relate the filter patterns to a behavioral outcome (e.g., keyword recognition). This technique is data driven—i.e., a computational approach (regression) is taken to estimate the relation between stochastically varying acoustic-speech features and listener responses (the classification image). For a complex stimulus such as continuous speech, this relation may be determined by multiple underlying factors—e.g., task demands, listener characteristics, response strategies, and perceptual acuity. As such, it can be difficult to interpret classification images, which (blindly) reflect the combined influence of these underlying factors. In this talk, I present the classification images from multiple speech experiments comparing (a) different stimuli (single- versus multi-talker), (b) different tasks (intelligibility versus emotion perception), and (c) different listener groups (persons with versus without hearing loss, musicians versus non-musicians) and discuss preliminary efforts to constrain the interpretation of classification images by appealing to the tenets of signal detection theory.

4:10

1pPP10. Multidimensional signal detection theory: Theoretical issues and empirical applications. Noah H. Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, French East 344, 3202 Eden Ave., Cincinnati, OH 45267, silbernh@ucmail.uc.edu)

Signal detection theory (SDT) provides a powerful set of tools for modeling choice data in a wide variety of domains. Fundamental to SDT is the distinction between noisy evidence and deterministic response selection. Multidimensional signal detection theory (MDSDT, aka general recognition theory) extends the basic concepts of SDT so that they are applicable to more complex tasks and data. Building on the SDT foundation of noisy evidence and deterministic response selection, MDSDT adds multiple ways in which dimensions may (or may not) interact. The additional complexity of multiple dimensions both broadens the range of applications of these tools and presents unique challenges. I will discuss the origins of MDSDT, a number of empirical applications of MDSDT in speech and hearing sciences, and some recent theoretical developments, focusing on model mimicry and identifiability and multilevel extensions of the basic model.

Contributed Papers

4:30

1pPP11. Regarding sources of irreducible variation in transformed up-down staircase experiments. Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 100 Exploration Way, Hampton, VA 23666, andrew.christian@nasa.gov)

A modicum of variation in the result is always expected when a psychoacoustic experiment is run multiple times on a single subject. Historically, this variation was attributed to human shortcomings: lapses in attention, inexperience, and the effects of memory, learning, and fatigue. It is demonstrated that the use of any psychophysical method that requires a binary response (i.e., a correct or incorrect response) will generate irreducible variation in the result—even for a perfect subject. At NASA, a recent battery of low-frequency simultaneous masking experiments was undertaken (see NASA-TM-2018-220120). This effort was preceded by a lengthy test design phase—also documented therein—during which it was determined that the test should consist of a large number of subjects, each to be tested minimally. For this, transformed up-down staircases were simulated in order to produce a prediction of the irreducible variance as a function of the staircases' parameters (e.g., length in the number of trials). The results from repeated test conditions taken during the experiment indicate that at least half of the variation observed within subjects may be coming from this irreducible source, though the propensity to produce inconsistent results may be a feature of some subjects more than others.

4:45

1pPP12. The invasion of psychoacoustics by inferential statistics. William Yost (ASU, P.O. Box 870102, Tempe, AZ 85287, william.yost@asu.edu) and M. Torben Pastore (ASU, Troy, NY)

The basic results that form the psychoacoustic foundation of our current knowledge of auditory perception, collected over nearly 50 years, were generated without any use of null-hypothesis significance testing (inferential statistics). The common use of inferential statistics in psychoacoustic articles published in JASA did not occur until the 1980s. Yet, today it is almost impossible to publish a psychoacoustic article in JASA, or anywhere

else, without using inferential statistics. In addition, recent controversies concerning replication in the social and behavioral sciences are beginning to encroach on the field of psychoacoustics. In this presentation, a case will be made that several elements of the experimental design inherent to many, if not most, psychoacoustic experiments often make the use of inferential statistics unnecessary. These aspects (many derived from the framework of the Theory of Signal Detection) include careful experimental control of stimuli and response acquisition, ratio-scale measurements, experienced subjects, large number of trials per subject, within-subject (repeated measures) experimental design, and strong theoretical context (e.g., an ideal observer). Some scenarios of experimental design and data presentation will be considered, suggesting that inferential statistical analysis might not be needed in some psychoacoustic research. [Work supported by NIDCD and Facebook Reality Labs.]

MONDAY AFTERNOON, 13 MAY 2019 STOPHER, 1:30 P.M. TO 3:55 P.M.

Session 1pSA

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Architectural Acoustics: Smart Materials for Acoustics and Vibrations II

Kathryn H. Matlack, Cochair
Department of Mechanical Science and Engineering, University of Illinois at Urbana-Champaign, 1206 W. Green St., Urbana, IL 61801

Bogdan Ioan Popa, Cochair
Mechanical Engineering, University of Michigan, University of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Invited Papers

1:30

1pSA1. Nonreciprocal metamaterial with programmable nonlinear virtual resistors. Amr M. Baz (Mech. Eng., Univ. of Maryland, 2137 Eng. Bldg., College Park, MD 20855, baz@umd.edu)

Emphasis is placed on the development of a class of active acoustic diodes and metamaterials in an attempt to control the flow and distribution of acoustic energy in acoustic cavities and systems. The proposed active nonreciprocal acoustic metamaterial (*ANAM*) cell consists of only one-dimensional acoustic cavity provided with active flexible boundaries. These boundaries are made from piezoelectric bimorphs with the inner layers which interact directly with cavity acting as sensors for monitoring the pressures of the propagating acoustic waves. The outer layers of the bimorphs provide the necessary control actions to an array of programmable nonlinear shunted resistors. These resistors are programmable and are designed in such a manner to introduce simultaneous nonlinear damping and cubic hardening stiffness effects. A lumped-parameter model of the *ANAM* cell is developed to control the nonreciprocal characteristics of the cell by the selection of the proper balance between the nonlinear damping and stiffness effects. Lyapunov stability criterion is used to generate the structure of such a balanced control strategy. The Harmonic Balance Method is used to predict the limit cycle behavior of the *ANAM*, the backbone characteristics and the stable limits of the control gains. Numerical examples are presented to demonstrate the effectiveness of the proposed *ANAM* in tuning and programming the directivity, flow, and distribution of acoustic energy propagating though the metamaterial.

1:50

1pSA2. Magnetorheological elastomer panels with reconfigurable magnetic mass for acoustic wave absorption. Carson Willey, Vincent W. Chen, Ken Scalzi (UES, Inc./Air Force Res. Lab., 2179 12th St., Bldg. 654/Rm. 304, Wright-Patterson AFB, OH 45433, carson.willey.ctr@us.af.mil), Philip Buskohl (Air Force Res. Lab., WPAFB, OH), and Abigail T. Juhl (Air Force Res. Lab., Wright-Patterson AFB, OH)

Lightweight metamaterials based elastomeric panels are used for sound absorption of low-frequency noise. It is beneficial to tune the resonant frequency of the metamaterial, by varying the stiffness of the panel, to enable use over a broader frequency range. The stiffness of magnetorehological elastomeric (MRE) panels, composed of iron particles embedded in a PDMS matrix, can be tuned by exposure to a magnetic field. This work represents an experimental realization and theoretical understanding of a magnetic mass decorated MRE metamaterial, which allows for tunable stiffness and easy reconfiguration of the masses applied to the panel. The circular panels are clamped at the rim, outfitted with neodymium magnets, and installed in an acoustic impedance tube, where they act as an acoustic barrier between the transmit and receive sides of the tube. There are three competing mechanisms that affect the sound transmission across the panel: (1) the mass increase associated with the magnetic masses, (2) the stiffening of the panel due to the magnetic field of each mass, and (3) the increased reflection from the placed masses. The experimental results demonstrate significant sound absorption by a thin metamaterial panel over a large frequency range through careful placement of magnetic masses.

2:10

1pSA3. Non-reciprocal sound propagation using acoustic resonators via space-time modulation. Junfei Li, Chen Shen (ECE, Duke Univ., 101 Sci.Dr, Rm. 3417, FCIEMAS, Durham, NC 27708, junfei.li@duke.edu), Xiaohui Zhu (ECE, Duke Univ., Harbin, Heilongjiang, China), and Steven Cummer (ME, Harbin Inst. of Technol., Durham, NC)

Breaking reciprocity is of fundamental importance in communication and signal processing and is less explored in the field of acoustics. Here, we show that non-reciprocal acoustic transmission can be realized in cascaded resonators that are periodically modulated. In the continuum limit, we also study the sound propagation in a continuously modulated effective medium. Functionalities such as mode conversion, parametric amplification, and phase conjugation are demonstrated. Finite-difference time-domain (FDTD) simulations are carried out to verify the results. An experimental platform is constructed which is composed of a waveguide system with side-loaded resonators. The back walls of the resonators are driven dynamically using speakers so that time modulation can be achieved.

2:25–2:40 Break

2:40

1pSA4. Design and fabrication of a linear broadband non-reciprocal acoustic waveguide using feedforward control. Nate Geib, Aritra Sasmal, Karl Grosh, Bogdan Ioan Popa, and Yuxin Zhai (Mech. Eng., Univ. of Michigan, 1587 Beal Ave. Apt 13, Ann Arbor, MI 48105, geib@umich.edu)

Acoustic metamaterials that exhibit non-reciprocal transmission have received substantial attention due to their wide range of applications such as noise control, diagnostic imaging, communications, and acoustic cloaking. Passive metamaterials achieve non-reciprocity through system resonances combined with nonlinearities or manipulation of phononic bandgap and are usually effective over a narrow band of frequencies. In this study, we describe the development of a new class of acoustic waveguides that exhibit linear non-reciprocal sound transmission over a wide range of frequencies. We use an open loop feedforward control mechanism where the measured local pressure is passed through an electronic controller and transmitted downstream to actuate an acoustic source. Analysis of the idealized one-dimensional model with a continuous distribution of probes and sources shows spatial asymmetry with attenuation in one direction and amplification in the other for frequencies ranging from one-half to five times the ratio of the active length of the segment to the acoustic wavelength. We show that such behavior can be replicated in a physical system with a finite array of acoustic probes and sources and develop the framework to compute the stability of this class of active waveguides. The experimental results for a simple feedforward model system are presented.

2:55

1pSA5. Non-reciprocity in mechanically modulated elastic metamaterials with geometric asymmetry. Benjamin M. Goldsberry, Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu)

Elastic metamaterials with time- and space-dependent effective material properties have received great attention as a means to generate non-reciprocal wave propagation in acoustic and elastic media. These materials have promise for applications such as acoustic communication devices with increased data throughput and improved vibration isolation components. One means to modulate the material properties of a heterogeneous medium is via nonlinear mechanical deformation on time scales that are slow compared to propagating waves. This approach has been explored by the authors using the finite element method (FEM) to demonstrate non-reciprocal elastic wave propagation in negative stiffness honeycombs with a time- and space-varying external pre-strain [Goldsberry *et al*., *JASA* **144**, 1763 (2018)]. One benefit of FEM is that it can be used to study the degree of non-reciprocity when varying sub-wavelength geometry or geometric modulations that are difficult or impossible to represent using analytical models. The present work employs FEM to investigate non-reciprocity in elastic lattices consisting of unit cells with varied geometric asymmetry and more general forms of mechanical modulation. [Work supported by National Science Foundation EFRI under Award No. 1641078.]

3:10

1pSA6. Design and experimental validation on acoustic valley Hall edge states in reconfigurable phononic elastic waveguides. Ting-Wei Liu and Fabio Semperlotti (Purdue Univ., 177 S Russell St., West Lafayette, IN 47907-2099, liutw@purdue.edu)

We report on the design and experimental validation of a tunable topological elastic lattice capable of supporting guided waves that are robust against back-scattering from disorder and defects. The topological lattice consists in a patterned aluminum thin plate having through-holes arranged in a hexagonal periodic configuration. The occurrence of the topological edge state is attributed to the acoustic analogue of the electron quantum valley Hall effect. By connecting two lattices having broken space inversion symmetry due to the application of opposite tunable strain fields, a topological transition emerges at the domain wall (i.e., the interface between them). Such a domain wall supports the formation and propagation of quasi-unidirectional edge states. The experimental validation of the topological waveguide is conducted on an aluminum plate that is already fabricated in a deformed (i.e., broken symmetry) configuration. The results confirm the existence of quasi-unidirectional edge states traveling along the domain wall and robust against sharp corners back-scattering. The experimental results also confirm the presence of mechanical energy flux vortices within the unit cell that are used to investigate the origin of the edge states robustness against disorder.

3:25

1pSA7. Architected hollow sphere foams for simultaneously tunable noise and vibration control. Yanyu Chen (Univ. of Louisville, 332 Eastern Pkwy, Louisville, KY 40292, yanyu.chen@louisville.edu)

Architected metamaterials, which are rationally designed multiscale material systems, exhibit novel functionalities and unique properties that cannot be readily achieved in natural bulk solids. In addition to unusual mechanical and physical properties, such as high specific stiffness and strength, negative Poisson's ratio, negative coefficient of thermal expansion, and negative compressibility, architected metamaterials have been designed and optimized for novel elastodynamic wave phenomena. One example of such architected metamaterials is phononic metamaterial, which consists of periodically topological structures and materials dispersions. These rationally designed structures enable the manipulation of propagating acoustic or elastic waves but not simultaneously. Here, we report a new type of architected hollow sphere foam that can attenuate sound and elastic wave synchronically. Our numerical simulation results suggest that the acoustic wave attenuation is controlled by the drilled area and thickness of the spheres, while the elastic wave propagation can be manipulated by changing the connectivity among the hollow spheres. We printed the architected hollow sphere foams and experimentally demonstrated the existence of omnidirectional acoustic and elastic wave bandgaps. Our findings reported here offer new opportunities to design lightweight architected metamaterials to simultaneously control undesired noise and vibration over a wide range of frequency.

3:40

1pSA8. Design of broadband active sound isolators based on acoustic bianisotropic metamaterials. Yuxin Zhai, Bogdan Ioan Popa, and Hyung-Suk Kwon (Mech. Eng., Univ. of Michigan, 3632 G.G. Brown Bldg., 2350 Hayward St., Ann Arbor, MI 48105, yxzhai@umich.edu)

The acoustic wave propagation through conventional materials is controlled by two parameters: the mass density and the bulk modulus. It has been shown that two additional parameters called Willis coupling terms are

needed in order to describe the dynamics of certain complex media called bianisotropic (Willis) materials. The few bianisotropic (Willis) materials reported so far have experimentally demonstrated the ability of these complex media to control the propagation of sound beyond what is possible with conventional media. For instance, it has been shown that bianisotropic materials could control reflection and transmission coefficients independently and enable high-efficiency sound beam steering devices. In this presentation, we will describe a previously unexplored property of acoustic bianisotropy. Namely, we will present a design of ultra-compact active bianisotropic metamaterials that can serve as excellent sound isolators. In addition, we will discuss the bandwidth and stability of our design. We will show that unlike conventional active sound control devices, active bianisotropic materials are stable even in dynamically changing environments without employing any instability mitigation algorithms.

MONDAY AFTERNOON, 13 MAY 2019 COMBS CHANDLER, 1:00 P.M. TO 5:15 P.M.

Session 1pSC

Speech Communication and Signal Processing in Acoustics: Exploring the Interface Between Linguistic Processing and Talker Recognition

Rachel M. Theodore, Cochair
University of Connecticut, 850 Bolton Road, Unit #1085, Storrs, CT 06269

Tyler K. Perrachione, Cochair
Department of Speech, Language, and Hearing Sciences, Boston University, 635 Commonwealth Ave., Boston, MA 02215

Chair's Introduction—1:00

Invited Papers

1:05

1pSC1. Listener sensitivity to structured phonetic variation. Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

Memory for spoken language is not a veridical representation of experience. Instead, memory reflects integration across our interlocutors' messages, resulting in robust memory for meaning with relatively poor memory for the specific form of the message. This is striking considering that in the process of mapping from speech to meaning, listeners show exquisite sensitivity to the acoustic-phonetic structure of speech. In this talk, I will review selected findings from work in our laboratory that examines listeners' ability to dynamically adapt to structured phonetic variation, focusing on variation associated with talkers' idiolects. These studies examine mechanisms that allow listeners to exploit structured variation for speech perception, voice recognition, and memory of spoken language. Collectively, the results (1) identify principles that govern how listeners modify the mapping to speech sounds to reflect cumulative experience with talkers' phonetic input, (2) show that sensitivity to structured phonetic input facilitates identification of a talker's voice in addition to the linguistic message, and (3) demonstrate that talker identity provides a critical structure for the integration of spoken language in memory. These findings help explicate a theoretical framework that accounts for tension in a linguistic architecture that uses both abstract and instance-specific representational knowledge to guide spoken language processing.

1:25

1pSC2. Recognizing speech in the context of talker variability. Shannon L. Heald and Howard C. Nusbaum (Psych., Univ. of Chicago, 5848 S. University Ave., B406, Chicago, IL 60637, smbowdre@uchicago.edu)

The relationship of the acoustic patterns of speech to phonetic categories varies across talkers. A change in talker therefore requires listeners use this relationship which depends on determining the acoustic-phonetic mapping for a new talker. The work from our group has demonstrated that tuning the perceptual system to the acoustic-phonetic system of a new talker is associated with a redirection of attention and a momentary increased load on working memory. These empirical findings support the hypothesis that speech perception is best thought of as an active cognitive process, where the ambiguity of the acoustic signal determines the mobilization of cognitive resources needed to support accurate perception. In this talk, I will (1) discuss the implication of these results for both cognitive and neural models of speech perception and (2) make the argument that extant theories of speech perception must address recognition in the context of acoustic-linguistic pattern variability, such as talker variability, as it is central to the computational problem of understanding speech perception.

1:45

1pSC3. Building talker knowledge from talker variability. Xin Xie (Univ. of Rochester, 850 Bolton Rd., Storrs, CT 06269, xxie13@ur.rochester.edu)

Listeners reliably extract invariant linguistic category information from speech despite massive cross-talker variability. Existing work shows that talker variability is in part handled by the learning and maintaining of category-specific acoustic distributions towards the statistics in the spoken input from a talker. *When* do listeners maintain information about talkers' speech after we meet them, and *what kind* of information should be maintained in principle? In this talk, I present a computational framework that addresses these questions by linking the statistics of the speech input (*production*) to predictions about *perception*. The results provide constraints on the type of inference underlying talker-related adaptivity during speech perception and have direct implications for current research on talker recognition.

2:05

1pSC4. Specificity and generalization in perceptual adaptation to systematic variation in speech. Lynne C. Nygaard and Christina Tzeng (Dept. of Psych., Emory Univ., Atlanta, GA 30322, lnygaar@emory.edu)

A signature problem in our understanding of spoken language processing is listeners' perceptual constancy in the face of considerable variability in the instantiation of linguistic form. Abundant evidence now suggests that listeners are sensitive to the fine-grained structure of linguistic units that signal differences among talkers and track and adapt to this structure during the perception of speech. This talk will present data from a perceptual learning paradigm addressing both the constraints and flexibility of speech perceptual mechanisms. The aim was to address the degree to which exposure to variation in and expectations about talker- and group-specific attributes influence the degree of specificity and generalization in the perceptual adaptation process. Listeners were exposed to talker-dependent variation in a linguistic form, and expectations about talker characteristics were manipulated. The results suggest that listeners dynamically adapt to and encode systematic changes in the linguistic category structure as a function of expectation and appear to flexibly integrate talker- and group-dependent sources of variation into linguistic representation and processing.

2:25–2:45 Break

2:45

1pSC5. Mutual predictability among speech sounds for talker adaptation and recognition. Eleanor Chodroff (Lang. and Linguistic Sci., Univ. of York, 2016 Sheridan Rd., Evanston, IL 60208, eleanor.chodroff@gmail.com) and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Baltimore, MD)

Processes of talker recognition and adaptation rely on a high degree of inter-talker phonetic variability and systematicity, respectively. While superficially in opposition, talker recognition in part depends on adaptation to the talker at hand. In this talk, we present evidence that talker variability is simultaneously extensive and structured within natural classes of speech sounds. In American English, talker mean peak frequencies for [s] span over 3000 Hz, but the variation in [s] is not independent of that in [z]: strong correlations of the talker mean peak frequency, among other phonetic dimensions, are observed between sibilant fricatives. Covariation among speech sounds indicates mutual predictability, such that evidence from one speech sound could be used to refine estimates or make predictions about a second. Listeners indeed demonstrate perceptual knowledge of covariation in generalized adaptation to novel talkers. After exposure to a talker with a relatively high- or low-peak frequency [z], listeners adjusted their [s]-[ʃ] boundary in accordance with the empirical covariation. As talker recognition entails estimation of a talker's phonetic parameters, prior perceptual knowledge of covariation could be used to refine estimation of multiple speech sounds from minimal exposure, thus accelerating processes of talker adaptation and recognition.

3:05

1pSC6. Who's talking and what are they saying: Phonetic cue distributions link speech and talker recognition. Dave F. Kleinschmidt (Psych., Rutgers NB, 152 Freylinguysen Ave., Piscataway, NJ 08854, dave.f.kleinschmidt@gmail.com)

On the one hand, talker variability is one of the fundamental challenges for speech recognition: each talker has their own mapping from linguistic units to sounds, which means that an effective listener must use a different recognition function for each talker. On the other hand, talker variability means that speech is a source of rich information about who the talker is. This dual nature of talker variability means that speech and talker recognition are inextricably linked: knowing something about who is talking makes it easier to understand what they are saying, and knowing something about how someone talks unlocks the rich social meaning of speech. I argue that the concept of a talker's generative model, or the probabilistic distributions of sounds associated with each phonetic/linguistic category, is a useful general purpose conceptual tool for understanding the link between talker variability, speech recognition, and social identity. With such phonetic cue distributions, we can use information theoretic tools to quantify both the extent and structure of talker variability across different phonetic systems and establish in-principle consequences of talker variability for both speech recognition and socio-indexical inferences from speech.

3:25

1pSC7. Asymmetries in phonetic memory across voices and contexts. Meghan Sumner (Dept. of Linguist, Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, CA 94305-2150, sumner@stanford.edu)

Listeners store highly detailed phonetic representations in memory. These representations are tied to talkers, whereby changing the talker negatively influences performance. This talk focuses on two related issues: (1) memory is not perfectly reliable and (2) the encoding of a single utterance across various voices and contexts is likely to vary. In this talk, I present the result from three experiments that highlight asymmetries that arise when we look the memory for spoken words in the continuous recognition paradigm when talker is a

between-subject variable, when talker is a within subject variable, and when talker is a within-subject variable in a divided attention condition. I show that memory for spoken words depends on talker voice, presenting talker voices together increases the differences found across talkers. These differences are larger again, under divided attention conditions. The phonetic memory for a single word depends on the talker, the social information processed about the talker by the listener, and stereotypes cued by the voices. This work focuses on the memory of words spoken by different talkers and is important in a broader context: Reports of linguistic experiences depend on linguistic memory, and here, we show that memory is both predictably unreliable and easily malleable.

3:45–4:05 Break

Contributed Papers

4:05

1pSC8. Toddlers' sensitivity to talker age in novel word learning: Children are better teachers. Yuanyuan Wang (Otolaryngology-Head and Neck Surgery, The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43202, Yuanyuan.Wang@osumc.edu) and Amanda Seidl (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

From early on, young children are sensitive to talker-specific attributes present in the speech signal. For example, infants attend selectively and learn better from their mother's voice than another female voice. Since age influences vocal quality, we asked whether toddlers show similar selective attention and learning from talkers of specific ages. We recorded visual and auditory stimuli from 3 adult males and females and 3 boys and girls. In Experiment 1, 24 toddlers viewed two side-by-side video clips of two talkers reciting a nursery rhyme and heard a soundtrack that matched with either the age or gender of one of the two talkers. The results revealed that children were able to match both vocal age and gender attributes to visual attributes. In Experiment 2, 24 toddlers were tested in a novel word learning task with talkers of different ages and genders. The results revealed that toddlers learned novel words from child talkers but failed to learn from adult talkers. These results suggest that young learners are sensitive to talker age information in speech and are biased towards learning words from younger talkers. We discuss the origins of such a bias.

4:20

1pSC9. Simulated cochlear-implant processing results in major loss of acoustic information regarding differences in talkers' voice qualities. Meisam K. Arjmandi, Hamzeh Ghasemzadeh, and Laura Dilley (Communicative Sci. and Disord., Michigan State Univ., Rm. 211A, 1026 Red Cedar Rd. Oyer Speech & Hearing, East Lansing, MI 48824-1220, khalilar@msu.edu)

The ability to recognize different voice qualities is essential for good talker identification; yet, little is known about how well voice quality cues of talkers are transmitted through the degraded speech signals delivered by cochlear implants (CIs). This study examined how CI speech filtering affects acoustic distinctiveness of individuals with and without voice disorder. Sustained /a/ vowels uttered by speakers with normal or disordered voice were processed using 4, 8, 12, 16, 22, and 32 channel noise-vocoders. The effect of CI processing on the distinctiveness of talkers with normal and disordered voices was measured using the Mahalanobis distance measure on Mel-frequency cepstral coefficients derived from samples across these groups. The analysis confirmed that CI vocoding dramatically degrades acoustic cues in frequency sub-bands that signal abnormal voicing behavior. Remarkable spectral degradation was observed in low- ($<$2 kHz), mid- (~4-12 kHz), and high-frequency ($>$12 kHz) bands for simulated conditions compared with unprocessed signals. These findings indicate that CI users likely have almost no ability to distinguish talkers differing in voice qualities. These results highlight challenges that CI users face for recognizing talkers differing in voice quality, due to these users' lack of access to fine-grained spectro-temporal details in voices.

Invited Papers

4:35

1pSC10. How task effects and reading ability relate to talker processing. Sandy Abu El Adas (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., NY 10012, sandyabu@nyu.edu) and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Studies show that listeners are better at processing talker information in their native language compared to an unfamiliar language. Several studies have explored two possible sources of this effect by manipulating lexical-semantic information on the one hand and phonological familiarity on the other. To probe these two types of information, researchers have manipulated the stimuli (e.g., phonologically similar or different languages, nonwords) and the listeners (e.g., listeners with reading impairments because reading ability is linked to phonological processing). These prior studies have found that individuals with poorer phonological processing also have poorer talker processing. However, it is unclear whether these individuals show poor performance in talker processing due to task demands related to creating talker representations in long-term memory. To test the question of task effects, we compare performance of individuals with a range of reading abilities on two tasks of talker processing: a discrimination task and an identification task. In addition, to control for both the lexical and phonological properties of the stimuli, stimuli were either words or nonwords and had either high or low phonotactic probability. Our preliminary results suggest that the effect of reading on talker processing depends on the type of task.

4:55

1pSC11. Talker variability, auditory attention, and speech processing efficiency. Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, tkp@bu.edu)

A classic finding in speech perception is that speech is processed more efficiently from a single, continuous talker than from mixed talkers. Given enormous variation in the acoustic realization of speech, it is thought that talker adaptation is necessary to ascertain the mappings between talkers' speech acoustics and listeners' abstract phonological representations. However, a suite of empirical studies

from our laboratory may suggest a predominately attention-based explanation for effects attributed to talker adaptation: using behavioral experiments, neuroimaging, and noninvasive brain stimulation, we have probed how speech processing efficiency under talker variability is affected by temporal, phonological, contextual, and expectational factors. We find that processing benefits from talker continuity are automatic and feedforward, depending on temporal continuity in the source of speech but not the amount of talker-specific phonetic detail or listeners' expectations about source continuity. Correspondingly, processing costs from talker discontinuity occur even when phonetic contrasts are unambiguous across talkers but are insensitive to the magnitude of phonetic variability, amount of preceding exposure to a talker, or top-down expectation about discontinuity. We consider how domain-general models of attention and auditory streaming may parsimoniously account for these differences in the speech processing efficiency between single and mixed talkers.

MONDAY AFTERNOON, 13 MAY 2019 MCCREARY, 1:30 P.M. TO 4:45 P.M.

Session 1pUW

Underwater Acoustics: Target and Radiation by Structures

Steven G. Kargl, Chair
University of Washington, Applied Physics Laboratory, 1013 NE 40th St., Seattle, WA 98105

Contributed Papers

1:30

1pUW1. Long-range detection of a spherical target located near the gas-saturated bottom. Natalie S. Grigorieva (St. Petersburg State Electrotech. Univ., Apt. 7, Krasnoputilovskaya St. 21, St. Petersburg 198152, Russian Federation, nsgrig@natalie.spb.su)

The study is devoted to modeling of the backscattering from a spherical target located near the gas-saturated bottom. Two cases are compared: when the sound speed in sediment is larger and smaller than the sound speed in water (rigid and soft bottom). The bottom is assumed to be a homogeneous attenuating fluid half-space. The transmitter/receiver is located in a homogeneous water half-space. Modeling is performed in the frequency band of 70–90 kHz for distances between the target and transmitter/receiver from 500 m up to 1 km. The spherical scatterer of a radius a is assumed to be acoustically rigid; $0.3 < a < 0.5$ m. For calculating the echo signal in the frequency domain, we have followed the Hackman and Sammelmann's general approach. The arising scattering coefficients of the sphere were evaluated with the use of the steepest descent method. The use of the obtained asymptotic formulas for the scattering coefficients allowed to decrease essentially a number of summands in the formula for the form function of the backscattered acoustic field. [Work supported by Russian Ministry of Educ. and Sci. under Grant 02.G25.31.0149.]

1:45

1pUW2. Canonical cylindrical target acoustic backscattering investigations. Joshua S. Davis (X11, Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City, FL 32407, joshua.davis1@navy.mil), Rodolfo Arrieta (X12, Naval Surface Warfare Ctr. Panama City Div., Panama City, FL), and Jermaine L. Kennedy (X11, Naval Surface Warfare Ctr. Panama City Div., Panama City Beach, FL)

In littoral environments where visibility can be low and objects of interest are often either partially or fully buried with respect to the water-sediment interface, the task of identifying unexploded ordinances (UXOs) is a challenging one. Downward looking sonar (DLS) systems have been shown to be capable of penetrating the seafloor of various littoral environments and therefore could be an effective tool for searching for UXOs. However, although DLS's can penetrate the seafloor, classification and identification (C&I) of targets of interest versus clutter are still a technical obstacle of interest. To address this, researchers at the Naval Surface Warfare Center Panama City Division (NSWC PCD) have been working on algorithms that can identify an object based on its acoustic backscatter response. The concept is similar to identifying the material composition of an object with optical spectroscopy but in this case using the acoustic response rather than the optical spectra. Initial investigations, analysis, and algorithm development have focused on the analysis of acoustic backscattering measurements conducted on various targets in the free-field using linear frequency modulated (LFM) interrogation pulses. Experimental measurements are compared with model results derived from a paper by Lecroq [1]. Data/model results as well as a process for potentially aiding automatic target recognition development will be discussed in this presentation. [1] F. Lecroq, L. Fernand, D. Décultot, and G. Maze, *J. Acoust. Soc. Am.* **91**, 1388 (1992).

2:00

1pUW3. Broad bandwidth acoustic backscattering from small fish : Measurements and cylindrical model. Henry M. Manik (Dept. of Marine Sci. and Technol., Bogor Agricultural Univ. (IPB) Indonesia, Marine Ctr. 3 FL Departemen ITK - FPIK IPB Kampus IPB Dramaga Bogor, Bogor 16680, Indonesia, henrymanik@ipb.ac.id)

Acoustic backscattering measurements were conducted of individual ray-finned fishes (*Gnathopogon caerulescens*) at frequencies from 30 to 150 kHz in an acoustic tank laboratory. Acoustic backscatter measurements for dead ray-finned fishes were made versus incidence angle from -30 deg to 30 deg relative to lateral aspect directions. Backscatter spectra from whole fish vary with the fish length and shift to lower frequencies at higher pitching angles. The backscatter spectra from pitching angle differ in both amplitude and positions. A cylindrical model of fish backscatter was developed and compared with the acoustic measurements.

2:15

1pUW4. Scattering of focused beams by spheres: Understanding the high-frequency angular structure. Timothy D. Daniel and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu)

Previous work on scattering of Bessel beams by spheres [P. L. Marston, *J. Acoust. Soc. Am.* **122**, 247–252 (2007)] and recent work expanding linear focused beams in terms of a Bessel beam superposition [T. D. Daniel *et al.*, *J. Acoust. Soc. Am.* **144**, 3076–3083 (2018)] provide a framework for exploring the scattering of focused beams by spheres. In analogy with the work done on the scattering of Gaussian beams by spheres [P. L. Marston, *J. Acoust. Soc. Am.* **129**, 1773–1782 (2011)], the angular scattering pattern of a sphere in a focused beam is calculated from a partial wave series with appropriate weighting factors (also known as beam shape coefficients). The high-frequency pattern for rigid spheres is found to have distinctive shadow boundaries in both the forward and backward scattering hemispheres. The sphere is centered on the beam axis. A physical optics model of the scattering reproduces the location and general shape of the shadow boundaries in both scattering hemispheres, confirming the calculated beam shape coefficients for a spherical cap focused source. [Work supported by ONR.]

2:30

1pUW5. Target strength measurements of spherical and wobbly bubbles. Alexandra M. Padilla and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Forest Park Apt. 281, Durham, NH 03824, apadilla@ccom.unh.edu)

Methane gas bubbles released from the seafloor transport gas upwards through the water column and in some cases even to the atmosphere. Researchers exploit the large acoustic impedance contrast between the gas within the bubble and the surrounding water to acoustically estimate the flux of methane in the ocean. Flux estimation employs the use of analytical acoustic scattering models to convert acoustic backscatter measurements of gas bubbles to size estimates. However, these models assume that bubbles are both spherical and that their radius is much smaller than the acoustic wavelength ($ka \ll 1$). Typically, bubbles in the ocean range from 1 to 5 mm in radius are non-spherical in shape and are observed for ka values that are greater than 0.1. A controlled shallow water tank (<6 m) experiment was conducted to assess the uncertainty associated with using analytical models that assume bubbles are spherical and that ka values are much smaller than one. Small, spherical and large, wobbly gas bubbles (radii ranging from 0.7 to 4.5 mm) were ensonified over a broad range of frequencies (10–250 kHz) in order to observe ka values from 0.1 to 5.

2:45

1pUW6. A focused lens for measuring object response to modulated radiation pressure. Timothy D. Daniel (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ivars P. Kirsteins (NUWC, Newport, RI), Ahmad T. Abawi (HLS, Inc., La Jolla, CA), and Philip L. Marston (Phys., WSU, Pullman, WA)

An ultrasonic lens was designed and tested for use in modulated radiation pressure experiments for the purpose of exciting and identifying the low frequency modes of objects in water. The lens was designed to produce a focused beam, and care was taken to minimize the spherical aberration introduced by the lens. This was done by carefully specifying the shape of the lens surface that was then machined using a computer-controlled lathe. The lens was placed in contact with an electrically excited piezoceramic plate. The quality of the focused sound field was tested and found to display properties predicted by simple models of the focal plane wave-field. The lens was then used to drive low-frequency modes of elastic objects in water using modulated ultrasonic radiation pressure in a way previously summarized involving a commercially made focused transducer [T. D. Daniel *et al.*, *J. Acoust. Soc. Am.* **140**, 3123 (2016)]. [Work supported by ONR.]

3:00–3:15 Break

3:15

1pUW7. Verification and validation of a coupled elasto-acoustic-damage system. Jonathan Pitt (AOE, Virginia Tech, 900 N. Glebe Rd., Arlington, VA 22203, jspitt@vt.edu)

A novel time-domain method for simulating dynamic damage evolution in a coupled structural-acoustic system is presented, with the aim of first principles modeling of acoustic emissions from failure. The system is derived via the theory of continuum damage mechanics and incorporates standard damage evolution models. The overall solution method is staggered, first solving for the dynamic damage evolution with an explicit step, and then using the new values of damage in the coupled computation of the structural-acoustic system. The spatial domain is discretized using a mixed finite element method, and the temporal domain is discretized with a higher-order implicit time discretization scheme. Code and solution verification of the fully coupled solution algorithm are presented, as are validation studies for cases with and without evolving damage. Applications with evolving damage and presents a methodology to study changes in the structural acoustic response from dynamically evolving damage in the structure are presented. Examples of downstream usage of the evolving structural response are discussed in the concluding remarks.

3:30

1pUW8. Sound scattering from targets in shallow water by finite element method combined with normal-mode method. Jinyu Li, Dejiang Shang, Jinpeng Liu, and Chao Zhang (Harbin Eng. Univ., 145 Nantong St., Nangang District, Harbin, Hei Longjiang 150001, China, jenny.mamie@163.com)

A hybrid method to calculate the sound scattering field of the target in waveguide is presented, combining the theory of the normal mode with the finite element method (FEM). Although FEM used only can provide accurate results, it is deeply restricted by the property of the computers and does not have a satisfying computing speed, for its calculation is closely related to the size and complexity of the target. The hybrid method presented in this paper can calculate the sound scattering field of more arbitrary and more complicated target over farther ranges in oceanic waveguide at low frequencies. First, separate the sound field into two parts, domain A, containing the target in the nearer range, and domain B meaning the rest domain of the field. In domain A, the sound field was calculated via FEM. Then, apply the normal mode theory to forecast the sound scattering field over farther ranges, using the pressure data on the intersection area between domains A and B obtained by FEM. The method has been used to calculate the sound scattering field of a rigid sphere in waveguide and based on the results, it is demonstrated that this hybrid method is accurate and efficient.

3:45

1pUW9. The acoustical characteristics of a metal ball entering into water. Dajing Shang and Qi Li (Underwater Acoust. Eng. Dept., Harbin Eng. Univ., Nangang District Nantong St. No. 145., Harbin City, Heilongjiang Province 150001, China, shangdajing@hrbeu.edu.cn)

The sound of objects into the water is very important to the detection and identification of underwater targets. The Metal ball into the water will produce an unique sound signal, which can be used for monitoring the target and evaluating the performance of the sonar. The main influence terms of the radiation noise of the metal ball into the water are analyzed in this paper, and the noise of the metal ball entering into water consists of initial impact sound and bubble sound, and the characteristics of the initial impact sound and the bubble sound are summarized. A set of metal ball entering the water equipment was set up to make the metal ball enter the water at different depths and angles, and the test was finished in the pool of the Harbin Engineering University Underwater Department. A special acoustic signal during the quiet area is discovered, which is related to the size of metal ball and the speed of the metal ball entering water. The sound of the metal entering water was measured, and the relationship between the sound characteristics of the metal ball entering water and the metal ball size, the velocity, and the angle of metal entering water was obtained by processing the measuring data.

4:00

1pUW10. The classification of spherical shells with varying thickness-to-radius ratios based on the auditory perceptive features. Xiukun Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China) and Yushuang Wu (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin 150001, China, wuyushuang1009@hotmail.com)

The detection and classification of underwater objects are an important part of the coast and harbor security. For silent targets, active sonar is always employed, and then, the information contained in the scattering echo is extracted and applied. Previous studies are aimed at targets with varied shapes, and the classification of objects with similar shapes has yet to be studied comprehensively. To address this issue, the classification of spherical shells with varying thickness-to-radius ratios (TRRs) is studied. The numerical results indicate that spherical shells can be roughly divided into three classes (extremely thin, thin, and thick) according to their scattering components. In this work, the scattering features, especially auditory perceptive features for spherical shells with different TRRs, are extracted and analyzed. Then, the features are imputed into a support vector machine classifier. The simulation results of the recognition rates under different features and signal-to-noise ratios will be present and analyzed. This study provides a novel concept for the classification of silent underwater targets.

4:15

1pUW11. The research of passive detection parameter measurement based on vector sensor. Juan Hui, Xianzhong Bu (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn), Anbang Zhao (Underwater Acoust., Harbin Eng. Univ., Harbin, HLJ, China), Dayu Wang, and Jin Li (The 54th Res. Inst. of China Electronics Technol. Group Corp., Shijiazhuang, China)

As one of the indispensable technologies for underwater target detecting, underwater acoustic passive detection technology has always been a hot spot in marine system research. A vector sensor can detect underwater targets by combination with the pressure and vibration velocity information; vector sensor array can obtain higher gain than pressure array with the same number hydrophones. Vector sensor can reduce the length of the underwater sensor array and can detect underwater targets in the case of low signal to noise ratio. This is the goal that many researchers strive to solve for many years; therefore, vector sensor passive detection technology has far-reaching scientific research value. Through the verification of the simulation experiment, the azimuth estimation technique and the adaptive frequency estimation technique are feasible. By estimating the azimuth sequence and frequency sequence of the target radiation signal, the relevant information of the target can be obtained, so it is of great significance in the research of underwater acoustic detection technology.

4:30

1pUW12. Acoustic radiation from elastic structures in shallow water by finite element method combined with normal modes. Buchao An, Dejiang Shang, Chao Zhang, and Yan Xiao (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, an_buchao@foxmail.com)

Acoustic radiation problem of elastic structures in shallow water has not been solved effectively. Finite Element Method (FEM) is not suitable for higher frequency and large range problems. In principle, the Combined Wave Superposition Method can deal with such problems. However, it is hard to optimize a large number of virtual sources. For low and middle frequency bands, we propose a new method combining FEM and normal modes. FEM is used to calculate the near range acoustic field of the structural source, then the eigen function expansion is performed on the field, and the coefficients of modes can be obtained by using the orthogonality of the eigen function. Therefore, we can calculate the acoustic field at any range. This method avoids the process of solving inverse matrix of a large complex matrix, which is an essential step in wave superposition method, and is very efficient to calculate the far field. To show the efficiency and accuracy of the method, the simulation results of various structural sources and waveguide models are compared with that of using FEM directly.

Additional registration fee required to attend this lecture.

MONDAY AFTERNOON, 13 MAY 2019 CLEMENTS, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Computational Methods for Describing Acoustic Propagation in Forests

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Computational methods for describing acoustic propagation in forests. Michelle E. Swearingen (U.S. Army ERDC/CERL, Construction Eng. Res. Lab., P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil)

Acoustic propagation through stands of trees, both large and small, can be markedly different from propagation through an open environment. Ground properties, meteorology, and significant scattering effects all play a role. An understanding of these effects, and computational methods used to describe them, can be utilized in applications ranging from noise mitigation to wildlife communication. This tutorial explores the myriad of computational methods that have been developed to describe acoustic propagation in forested environments. Beginning with the early simple empirical models describing attenuation, the presentation then narrows the field of view to contributions of individual components within a forest. Models for describing scattering by trunks and foliage, both individually and as ensembles, are presented. Next, the integration of these individual components into the fully coupled system that includes ground properties and meteorology, within computational methods such as the parabolic equation (PE) and finite-difference, time-domain (FDTD) methods are shown. Finally, examples are presented showing how computational methods for forest acoustics can be used for evaluating noise mitigation strategies and wildlife studies.

MONDAY EVENING, 13 MAY 2019 WILSON ROOM, 5:00 P.M. TO 6:15 P.M.

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

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

R. J. Peppin, Vice Chair ASC S2
5012 Macon Road, Rockville, MD 20852

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and four of its subcommittees, take note that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

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

Session 2aAAa

Architectural Acoustics and Noise: Libraries, Media Centers, and Similar Spaces

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

Damian Doria, Cochair
Stages Consultants LLC, 75 Feather Ln., Guildford, CT 06437-4907

Chair's Introduction—8:00

Invited Papers

8:05

2aAAa1. Introduction and overview—Libraries as a building type and reflections of multipurpose trends that have an acoustical impact. Dennis Paoletti (Paoletti Consulting, 708 Foothill Dr., San Mateo, CA 94402, dpaoletti88@gmail.com)

Libraries have developed from vast storehouses of bookshelves and hard covered manuscripts and literary materials to vibrant community centers and true multipurpose facilities, acoustically. This paper will discuss the: history of libraries, politics of libraries, architectural design of libraries, acoustical design of libraries, and the future of libraries. Examples of library projects ranging from main urban public libraries and university library systems to a number of smaller, standalone local community libraries. will provide interesting insight into the variety of activities that go on today in libraries that challenge the best of our acoustical design sensibilities and controls. Libraries share the basic acoustical parameters of many other building types, e.g., open plan spaces, offices and conference rooms, cafes, and even digital/multimedia and performing arts facilities. Lessons learned from challenging consulting efforts, especially when budgets are limited, will be explored and discussed.

8:25

2aAAa2. Soundscape of the evolving library. Gary W. Siebein, Hyun Paek, Marylin Roa, Keely Siebein, Jennifer R. Miller, Gary Siebein, and Matthew Vetterick (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Historically libraries were thought of as places of quiet reading and research. There were 2 general acoustical design approaches to reading rooms in libraries. One was to use large volume spaces with sound reflective materials that amplified every sound made in the room so that people would be self-conscious of the sounds that they made. A second approach was to design rooms with sound absorbent finishes that would reduce the spread of sounds from one part of the space to another. The evolving library of the future has many more complex functions and rich programs than the traditional library. Many of these encourage discussion, recreation, public involvement, experiencing multi-sensory media in addition to traditional books and a variety of other activities such as coffee shops, community meeting rooms, auditoriums, recording studios, listening rooms for amplified audio, interactive computer work stations, collaborative work/study areas where students can gather, talk, view and listen in active sessions all in close proximity to each other. Case studies of 2 libraries will be presented to identify links between soundscape design issues, architectural planning strategies and practical systems for designing the library of the future as it continues to rapidly evolve in concept.

8:45

2aAAa3. Recording room trends in libraries. Nicole Cuff (Acentech, 33 Moulton St., Cambridge, MA 02138, ncuff@acentech.com)

The use of libraries in communities and universities continue to evolve to where now there is a trend toward recording rooms in public and university libraries. One such space is a recording suite in a public library intended for the community to use, teens to seniors, with fully amplified bands next to study spaces, with control equipment available for patrons to use. Another in a private university is a podcast suite that overlooks a 4-story glass Atrium, which will be the new central meeting place on campus. The author even personally used recording studio space immediately adjacent to quiet study space at a private university library with their singing group to record an album. Some of the strategies used to make these spaces work together acoustically will be discussed.

9:05

2aAAa4. Acoustic conditions in two libraries at the University of Nebraska. William Spallino (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S 67th St. #210C, Omaha, NE 68182, wspallino@huskers.unl.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Lincoln, NE)

University libraries serve many roles for students, from group gathering places to quiet study spaces; this makes analyzing and optimizing their acoustics very worthwhile. This paper reviews the acoustics of both main areas and small-group study rooms at the University of Nebraska—Omaha's Criss Library and the University of Nebraska—Lincoln's Love Library. Sound level meters were used to log sound levels in the two libraries over multiple days. Those data were then analyzed to understand the libraries' soundscapes through metrics like average sound levels, statistical sound levels, and occurrence rates of specific sound levels. Impulse responses and transmission loss of partitions were also measured. From those analyses, conclusions are drawn about the acoustic behavior of the libraries and its appropriateness for their purpose.

9:25

2aAAa5. West Hollywood Library—More than just another pretty space. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The West Hollywood Library is a 35 000 sqft branch of the County of Los Angeles Library, with construction funded by the City and operated by the County. It serves as a cultural and artistic community center, and includes a children's library, teen center, career development center, special collections, café, bookstore, two parking garages, community meeting room, and multipurpose public meeting room (for council and commission meetings, performances, and other uses). An ornamental wood ceiling, the wood stage in the multipurpose meeting room, the children's reading room, and the overall aesthetic presented interesting challenges for acoustics. Visual aesthetic and acoustic performance aspects will be discussed, along with references to various other library projects.

9:45

2aAAa6. Acoustics and AV technology of modern collegiate learning centers. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and William Roland (Roland, Woolworth & Assoc., Meridian, MS)

Many universities are creating new dedicated spaces for interactive learning that can host all subjects and a more modern and holistic approach to collegiate education. In order to achieve high performance facilities that mix classrooms, testing rooms, collaborative spaces, offices, and counseling all together, acoustical and audio visual requirements must be coordinated early in the project with the end user, with room to adjust to changing programming and technology. This paper presents 3 case studies of repurposed and new collegiate construction and outlines the types of program requirements and the approach to acoustics and audio visual for these projects.

10:05–10:20 Break

10:20

2aAAa7. Acoustical design for libraries in schools to comply with ANSI S12.60. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

In K-12 schools, the library functional space is now usually called the media center, as this space houses and disseminates information that is stored on both print and digital media. These spaces have fewer sound absorbent book stacks than were found in previous print-only libraries. Instead, computer workstations for student access to local and web-based material are provided. Often these facilities are used as group learning spaces, with several group classes occurring simultaneously. These changes to the traditional school library space present challenges to the acoustical design. Another challenge is that in a number of school districts across the US, a media center must be designed to meet the requirements of ANSI/ASA 12.60-2010/Part 1, "American National Standard Acoustical Performance Criteria, Design Requirements and Guidelines for Schools, Part 1: Permanent Schools," also known as the "Classroom acoustics standard." This is because a media center can be defined as a core learning space that is subject to that standard, with its requirements for low noise and reverberation, and for higher sound isolation from adjacent spaces. Several case studies are presented which highlight the design challenges for modern school media centers, and the successful design solutions.

10:40

2aAAa8. Considerations for media center design in unconventional environments. Melvin Saunders (None, 1601 Elm St., Fl. 33, Dallas, TX 75201, melvin.saunders@saundersassoc.com)

The consolidation and simplification of media in production environments has rapidly changed the way that many businesses develop content. Rather than outsourcing media development, many clients have undertaken the task to develop content for both internal and external consumption by building in-house media centers. This paper will focus on some of the lessons learned for these types of projects from both a remediation and new construction process. Acoustical solutions for projects of varying budget and scope including room acoustics, sound isolation, and HVAC noise abatement will also be explored.

11:00

2aAAa9. A church space that could be a library, media room, meeting room, performance space, or dining room—or with proper room acoustic modifications—An effective sound reinforcement system including audio recording, and suitable video recording and projection it could function as all of the above noted spaces. Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, bob@rccoffeen.net)

In a nearby church that was recently visited a space was viewed that is nicely appointed from an aesthetic viewpoint but primarily due to room acoustics, noise issues, and lack of proper sound and video systems, it did not seem to adequately serve any of the functions noted by the title of this presentation. The church is considering a renovation of this space. This presentation will describe how the space can serve all of the noted functions with relatively simple and easy to operate acoustic, sound system, and video modifications without significantly disturbing its current and desirable appearance.

11:20

2aAAa10. Rethinking libraries—Three case studies. Evelyn Way (Stages Consultants LLC, Chicago, IL) and Damian Doria (Stages Consultants LLC, 75 Feather Ln., Guildford, CT 06437-4907, damianjdoria@gmail.com)

As the nature of library usage changes, traditional book stacks and individual workspaces are giving way to group activities, informal meetings, classes, and makerspaces. In the past, distraction was managed not through the acoustic environment, but operationally through behavior enforcement and having separate physical spaces for different activities. Now excessive reverberation and inappropriate background noise levels significantly inhibit the usability of a library. Conversations from the café must not disturb small group work areas or instruction on the three-dimensional printer, community meetings need to happen at the same time as classes on digital photo processing, and childrens' story time occurs while the local entrepreneur researches how to register their business. We examine the acoustic design for a community library as collaboration center, a community library as recreation center, and a school library integrated into the common space.

11:40

2aAAa11. Acoustic design of today's library. John C. Swallow and Michael J. Wesolowsky (Swallow Acoust. Consultants Ltd./ThorntonTomasetti, 23 - 366 Revus Ave., Mississauga, ON L5G 4S5, Canada, jswallow@thorntontomasetti.com)

Modern libraries are very different from past libraries, offering multiple services and having wildly different acoustic requirements. Reviewed are new uses and functional requirements: more than just a traditional information resource, the library features information sharing, collaboration, dissemination and production as a media centre. A brief history of library design is provided. Over decades, library design philosophy has evolved which has provided its own challenges. Modern building design presents more challenges; adding in the new users and uses—which even vary by age group, the acoustic designer now faces enormous challenges. Acoustic design measures are reviewed and found wanting: new measures are needed. Case studies show the variety of uses, novel design solutions and highlight how a detailed knowledge of actual users and uses is required to allow all the functions to more-or-less comfortably co-exist.

TUESDAY MORNING, 14 MAY 2019 EXHIBIT HALL, 8:00 A.M. TO 12:00 NOON

Session 2aAAb

Architectural Acoustics: Student Design Competition (Poster Session)

David Woolworth, Cochair
Roland, Woolworth & Associates, 365 CR 102, Oxford, MS

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

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund, the Wenger Foundation, and the National Council of Acoustical Consultants is sponsoring the 2019 Student Design Competition that will be professionally judged at this meeting.

The competition involves the design of a new municipal building including a court room and a community hall.

The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of USD$1600 will be made to the submitter(s) of the design judged "first honors." Four awards of US$800 each will be made to the submitters of four entries judged "commendation."

Session 2aAB

Animal Bioacoustics and Signal Processing in Acoustics: Bioinspiration and Biomimetics in Acoustics I

Rolf Müller, Chair
Mechanical Engineering, Virginia Tech, ICTAS Life Sciences District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061

Chair's Introduction—8:30

Invited Papers

8:35

2aAB1. Most animals hear acoustic flow instead of pressure; we should too. Ronald Miles (SUNY Binghamton, Dept. of Mech. Eng., Vestal, NY 13850, miles@binghamton.edu)

The majority of animals that hear sound do so by detecting the minute fluctuations in the velocity of the medium. They do this by sensing the deflection of thin hairs that are driven by viscous forces in the acoustic medium. This is in contrast to the detection of sound pressure, as is used in some animals, including humans. Nearly all microphones respond to sound through the use of a thin diaphragm or membrane designed to experience a net fluctuating pressure in a sound field. Here, we explore technologies for achieving precise detection of sound using a mechanical structure that is driven by viscous forces associated with the fluctuating velocity of the medium. In one example, this has been shown to result in a directional microphone with flat response from 1 Hz to 50 kHz [Zhou *et al*., "Sensing fluctuating airflow with spider silk," *Proc. Natl. Acad. Sci*., 201710559(2017)]. Methods of creating extremely compliant, viscous-driven velocity sensors are discussed along with a new technique for capacitive transduction to achieve an electronic output.

8:55

2aAB2. Animals can measure acoustic flow; with the help of MEMS, we can too. Stephane Leahy and Ahmed Abdelaziz (Soundskrit, 780 Ave. Brewster, Ste. RC-016, Montreal, QC H4C 2K1, Canada, stephane.leahy@soundskrit.ca)

Animals measure acoustic flow by using thin hairs. These hairs are essentially mechanical structures that are driven by viscous forces. Manufacturing thin hair-like structures is challenging, but we can think of other mechanical structures driven by viscous forces that can be manufactured more easily. Here, we discuss how MEMS technology can be used to fabricate a thin porous membrane and electrodes for measuring acoustic flow. We discuss the design trade-offs between performance, manufacturability, and product integration. For example, from a performance perspective, it is desirable that the membrane be as thin as possible, but from a manufacturing perspective, it is easier to make it thicker. We present preliminary results on early prototypes and discuss avenues to create a MEMS acoustic flow sensor that could be found in your smartphone within the next few years.

Contributed Papers

9:15

2aAB3. Emission baffle deformations in bat biosonar and biomimetic systems. Liujun Zhang (ME, ICTAS II, Virginia Tech, 1075 Life Sci. Circle, Blacksburg, VA 24060, sdujune@gmail.com), Luhui Yang (Shandong Univ., Jinan, China), and Rolf Müller (Virginia Tech, Blacksburg, VA)

Old-World leaf-nosed bats (Hipposideridae) and horseshoe bats (Rhinolophidae) are two families of echolocating bats that emit their biosonar pulses through nostrils which are surrounded by elaborated baffle shapes ("noseleaves"). Prior work has shown that the noseleaves change shape in synchrony with ultrasound emission. These deformations involve at least two noseleaf parts with static (geometric) as well as dynamic complexity. To investigate how these deformations could be used in bioinspired systems, data has been collected from bats recorded with microphone/camera arrays as well as a biomimetic sonar head with a biomimetic noseleaf model inspired by Pratt's roundleaf bats (*Hipposideros pratti*). Both sets of experiments have produced qualitatively similar results that show an emitted wave field which depends on frequency, direction, and also time. Furthermore, the results obtained with the biomimetic noseleaf have demonstrated that noseleaf deformations during ultrasound emission can result in an enhanced information-encoding capacity. Current work is aimed at establishing whether this enhanced coding capacity can be utilized to improve performance, especially in sensing tasks that are related to the natural environments in which the bats' biosonar systems operate. It also needs to be established how much detail needs to be mimicked in an engineered system to harness these effects.

9:30

2aAB4. Hipposideros pinna motion patterns as an inspiration for dynamic biomimetic sensing. Pennine Jiu, Anorexia Yin, and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Circle, Blacksburg, VA 24061, peiwenq@vt.edu)

Many bat species, e.g., among the horseshoe bats (Rhinolophidae) and Old-World roundleaf-nosed bats (Hipposideridae), have highly mobile pinnae. Experiments with biomimetic reproductions of the pinnae shapes/motion patterns in these species have demonstrated a dynamic enhancement of the performance in acoustic direction-finding paradigms. It could hence be hypothesized that reproducing the full range of pinna mobility patterns in bats will allow realizing the full range of the animals' biosonar capabilities. Pinna motions in rhinolophid and hipposiderid bats have been shown to fall into two distinct categories, being either rigid rotations (i.e., changes in orientation but not in the shape itself) or deformations that change both pinna orientation and shape. To characterize the variability in the rigid rotations, landmarks on the pinnae of hipposiderid bats have been tracked using stereovision. From this kinematics data, an axis-angle representation of the pinna rotations was estimated. The results showed that the axes of pinna rotations can scatter widely in orientation, covering a range of directions that extends 180 degrees in azimuth and 180 degrees in elevation, about 40 times larger than the estimated error of the employed method. Hence, biomimetic reproductions of bat ear mobility should explore how to make use of such a variability.

9:45

2aAB5. Reducing bats' pinnae deformation complexity for a biomimetic reception baffle dynamics. Jia Guo, Andrew Kurdila, and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., 144 Durham Hall, 1145 Perry St., Blacksburg, VA 24060, jguo18@vt.edu)

Horseshoe bats are known to have about 20 muscles on each pinna that allow the animals to deform their pinnae during echo reception. Simplified biomimetic reproductions of this reception baffle dynamics have demonstrated the encoding of additional, useful sensory information. However, reproducing the full complexity in the actuation of the bat pinna shapes by virtue of numerous muscles poses a daunting challenge. Furthermore, it remains unknown which level of complexity is necessary to realize the functional advantages of the bats' pinna dynamics. To address this issue, the current work has focused on non-rigid deformations of the pinnae. 3D kinematic data for a dense grid of landmark points distributed over the pinna surface has been used as input for manifold learning algorithms designed to reduce the dimensionality of the motion. The results of this analysis suggest that the deformations of the pinna can be described in spaces that with much lower dimensionality than the original kinematic data. In addition, a kernel method has been implemented to generate a reduced model of the pinna deformations. Ongoing work is aimed at understanding how these low-dimensional descriptions of the pinna deformation relate to muscular coordination patterns in the pinna and its dynamic functional characteristics.

10:00–10:15 Break

10:15

2aAB6. Fast moving biomimetic ears. Xiaoyan Yin and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir., Blacksburg, VA 24061, xiaoyan6@vt.edu)

Like many other mammals, bats are known to rely on spectral signatures to determine the directions of incoming sounds, especially in elevation. Because of its simple hardware and computational realizations, this principle could also be suitable for small, parsimonious biomimetic sonar systems. However, the biosonar systems of bat species in families such as the rhinolophids and hipposiderids often—but not always—concentrate most of the emitted energy within a narrow frequency band in order to pick up prey-induced Doppler signatures. It remains unclear how the use of such narrowband biosonar pulses could be reconciled with direction-finding based on spectral signatures. A possible solution to this paradox could be provided by fast pinna motions in these animals that have been shown to produce readily perceivable Doppler-shift signatures in biomimetic reproductions. In the current work, such a biomimetic pinna with fast deformations has been used to map Doppler-shift signatures as a function of direction. For this purpose, the Doppler-shift signatures were clustered based on the similarity of their spectrogram-representations. Mapping the signatures' different cluster associations into direction space resulted in contiguous patches. Hence, it should be possible to obtain stable estimates of target direction based on the received Doppler signatures. <audio controls = "controls" style = "display: none;" > </audio> <audio controls = "controls" style = "display: none; " > </audio>

10:30

2aAB7. Coordination of sonar emitter and receiver dynamics inspired by bats. Shuxin Zhang (Virginia Tech Int. Lab.,School of Phys., Shandong Univ., Shanda South Rd. No. 27, Jinan, Shandong 250100, China, shuxinsduvt@yahoo.com), Yanming Liu (Key Lab. of High Efficiency and Clean Mech. Manufacture, School of Mech. Eng., Shandong Univ., Jinan, Shandong, China), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Recent biomimetic sonar systems have started to mimic the mobility seen in the emission and reception baffles of bats (i.e., "noseleaves" and pinnae) in greater detail. Some of these efforts have included rigid rotations as well as non-rigid shape changes. Since some bat species are known to actuate their noseleaves during pulse emission and their two pinnae during pulse reception, it may be hypothesized that coordination between the dynamic behaviors of these baffle structures is important to system-level biosonar performance. However, it remains to be determined how the complicated rigid and non-rigid changes to the noseleaves and pinnae of bats are coordinated in the animals' biosonar behaviors. To shed some light on this dynamic system integration in bat biosonar, we have conducted experiments with a species of hipposiderid bat (Pratt's roundleaf bat, *Hipposideros pratti*) where at least 24 landmark points have been placed on the noseleaf and both pinnae in order to track the non-rigid shape changes in all these three structures simultaneously. A canonical correlation analysis has confirmed that the motions of the noseleaf and both pinnae are highly correlated with each other. Work to identify the nature of these relationships using techniques from nonlinear dynamics is currently underway.

10:45

2aAB8. Pulse-train time structure for dynamic biomimetic sonar. Yanming Liu (Key Lab. of High Efficiency and Clean Mech. Manufacture, School of Shandong Univ., Shan Dong Univ., 27 Shanda Nanlu, Jinan 250100, Shandong 250100, China, larry.young.ming@gmail.com), Shuxin Zhang (Virginia Tech Int. Lab., School of Phys., Shandong Univ., Jinan, Shandong, China), and Rolf Müller (School of Mech. Eng., Virginia Tech, Blacksburg, VA)

The biosonar pulse trains of echolocating bats can have pronounced time patterns. A well-known example are adaptive increases in pulse repetition rates seen when certain bat species approach their prey. Furthermore, the production of pulse groups has been reported to create pulse-timing diversity outside of prey pursuit in several bat species. In order to replicate the biosonar capabilities of bats, a biomimetic sonar system may have to integrate this level of adaptivity. If the system also mimics the peripheral dynamics, i.e., shape deformations of the noseleaves and pinnae, in species such as the horseshoe bats, the question arises how the pulse timings and the shape deformations should be best coordinated. To investigate this issue, synchronized recordings of noseleaf motion and pulse emission patterns have been obtained from hipposiderid bats. The noseleaf deformations have been characterized (clustered) based on reconstructed trajectories of five landmark points. Similarly, the corresponding biosonar pulse trains were classified based on similarity metrics developed for neural spike trains and are based on cost assigned to that operations that are needed to be performed to transform one spike pattern into another. The relationships between distances in the noseleaf motion and pulse-train domains are the subject of ongoing research.

11:00

2aAB9. Biosonar inspiration for radar waveform design. Gregory E. Coxson (Elec. and Comput. Eng., U.S. Naval Acad., M/S 14B, 105 Maryland Ave., Annapolis, MD 21402, gcoxson@ieee.org)

Bat biosonar offers a natural source for biomimetic design of radar waveforms with inspirations falling into two categories: (1) biosonar principles similar to ones already employed in radar and (2) principles used by bats that operate in ways not yet understood or not yet embraced yet for radar. This second type offers the possibility for driving radar innovation. Hyperbolic frequency modulation (HFM) waveforms were one of the first aspects of bat biosonar to catch the attention of the radar community. Relative to the popular linear frequency modulation (LFM) waveform, HFM is less vulnerable to Doppler shifts from targets' relative velocities. Another promising possibility for biomimetics is the remarkable ability of some bats to resolve closely-spaced objects using acoustic frequencies on a small platform. Bats' ability to operate in swarms is of interest to radar designers concerned about mutual interference between ships or planes using the same waveforms and frequencies. As bat researchers approach an understanding of the mechanisms behind these, and other, abilities of bats, their results will find a ready audience. This talk will discuss aspects of bat biosonar offering payoffs for radar design, including but not limited to improved resolution, Doppler tolerance and mitigation of inter-system interference.

11:20

2aAB10. Cochlea-inspired sonar signal processing. Bryan D. Todd (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City Beach, FL 32407, Bryan.D.Todd@navy.mil) and Rolf Müller (Virginia Tech, Blacksburg, VA)

Many attempts have been made to model the signal transformations that occur in the mammalian cochlea, in particular with respect to modeling human hearing. Cochlea models that have been developed in the last few decades fall into several categories, e.g., electrical analogs, digital filter models, and incorporate different levels of complexity from entirely linear to strongly nonlinear. Similarly, the implementations used have included various numerical approaches as well as analog and digital hardware architectures. Mammalian hearing systems are known to be highly capable when it comes to analyzing sound from complex natural environments and soundscapes. It may hence be hypothesized that the signal transformations that occur in the cochlea play a critical role in facilitating these capabilities since the cochlea is positioned at the critical first stage of auditory signal encoding. The goal of the research presented here is to evaluate the signal transformations of the mammalian cochlea in the context of sonar signal processing, especially for automatic target recognition in difficult environments such as reverberant shallow water. The cochlea of bats could proof a suitable model for these applications since many bat species are known to be capable of solving demanding sonar tasks under difficult conditions and cluttered environments.

Session 2aBA

Biomedical Acoustics and Signal Processing in Acoustics: Cardiovascular Ultrasound: Imaging and Therapy I

Kevin J. Haworth, Cochair
University of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

Jonathan A. Kopechek, Cochair
University of Louisville, 2301 S Third St., Lutz Hall, Room 400, Louisville, KY 40292

Invited Paper

8:30

2aBA1. Left ventricle blood flow patterns in a mouse model of temperature sensitive sodium channelopathy. Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10006, jkettermeister@gmail.com), Orlando Aristizabal (Skirball Inst. of Biomolecular Medicine, NYU School of Medicine, New York, NY), Akshay Shekhar (Leon H. Charney Div. of Cardiology, New York Univ. Langone Health, New York, NY), Billy Y. Yiu (Dept. of Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada), Colin K. Phoon (Div. of Pediatric Cardiology, Hassenfeld Children's Hospital at NYU Langone, New York, NY), Glenn I. Fishman (Leon H. Charney Div. of Cardiology, New York Univ. Langone Health, New York, NY), Alfred C. Yu (Dept. of Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada), and Ronald H. Silverman (Dept. of Ophthalmology, Columbia Univ. Medical Ctr., New York, NY)

Sophisticated methods of visualizing and analyzing the complex blood flow patterns in humans have been available for a number of years on ultrasound machines. To date, these methods have not been applied to healthy mouse models or mice with cardiac defects. Here, a high-speed plane-wave imaging approach was used to study left ventricle blood flow patterns in a model of sodium channelopathy using mice lacking fibroblast growth factor homologous factor 2 ($Fhf2^{KO}$) and littermate wild-type (WT) controls. FHF2 binds to the cytoplasmic tails of voltage-gated sodium channels in mouse cardiomyocytes and modulates channel inactivation and cellular excitability. $Fhf2^{KO}$ mice have impaired cardiac sodium channel function that when body temperature rises causes a severe reduction in cardiac sodium currents, cardiomyocyte excitation, and conduction failure. The effect is reversible as body temperature returns to normothermia. $Fhf2^{WT}$ and $Fhf2^{KO}$ mice were imaged with a Verasonics Vantage 128 using an 18-MHz linear array with a 30 kHz absolute plane-wave transmission rate. The mice were supine on an imaging platform and a warm-air source was used to raise the body temperature. Surface ECGs were continuously recorded throughout the duration of the experiment. Data were post-processed using a least-squares, multi-angle Doppler analysis approach to obtain vector-flow estimates at all pixel locations. Vortex patterns, flow rates and ECG signals were compared between the $Fhf2^{WT}$ and $Fhf2^{KO}$ mice.

Contributed Papers

8:50

2aBA2. Live color encoded speckle imaging platform for real-time complex flow visualization *in vivo*. Billy Y. Yiu (Elec. and Comput. Eng., Univ. of Waterloo, 250 Laurelwood Dr., Waterloo, ON N2J0E2, Canada, billy.yiu@uwaterloo.ca), Mateusz Walczak, Marcin Lewandowski (Inst. of Fundamental Technol. Res., Warsaw, Poland), and Alfred C. Yu (Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

Complex flow patterns carry valuable diagnostic information but they cannot be visualized in a time-resolved and intuitive manner in real-time. In this work, we present the first real-time scanning platform for a high frame rate ultrasound technique called color encoded speckle imaging (CESI) and its use in visualizing blood flow dynamics *in vivo*. CESI visualizes these dynamics through duplex rendering of flow speckle motion and color-coded Doppler velocity estimates. Its live implementation was achieved by integrating a 192-channel programmable ultrasound front-end module, a 5 GB/s capacity data streaming link, and a series of computing kernels implemented on the graphical processing unit (GPU) for beamforming, Doppler processing and display rendering. A slow-motion replay mode was also included to offer coherent visualization of CESI frames acquired at sub-millisecond resolution (0.3 ms). The live CESI scanning platform was found to be effective in facilitating real-time image guidance (>20 fps). *In vivo* pilot trials also showed that live CESI, when operating in replay mode, can temporally resolve the formation and dissipation of recirculation at the carotid bifurcation and can reveal flow dynamics in the brachial vein during a fist-clenching maneuver. Overall, live CESI has potential for use in routine *in vivo* investigations that seek to identify complex flow dynamics in real-time and relate these dynamics to vascular pathophysiology.

9:05

2aBA3. Design of carotid bifurcation phantoms for integrative imaging investigations of arterial wall and flow dynamics. Adrian J. Chee, Billy Y. Yiu, and Alfred C. Yu (Univ. of Waterloo, 250 Laurelwood Dr., Waterloo, ON N2T 2S1, Canada, adrian.chee@uwaterloo.ca)

Vascular phantoms are well regarded as essential experimental tools in the development of new ultrasound techniques for assessing wall mechanics and blood flow. However, existing phantoms are ill-suited for evaluation of integrative imaging methods that seek to concurrently assess biomechanics and hemodynamics. Here, we present a novel design protocol for acoustically-

compatible anthropomorphic walled phantoms with artery-like vessel elasticity of carotid bifurcation with stenotic-narrowing. Our protocol involved a set of three-dimensional printed mold parts (consisted of a vessel core and an outer mold) for investment casting of polyvinyl alcohol solution to construct the elastic vessel tube. Agar-gelatin slab was formed around the vessel tube mimicking surrounding tissue. For demonstration, a set of healthy and stenosed (25%, 50%, and 75%) carotid bifurcation phantoms were developed. Imaging experiments were performed on these phantoms to visualize complex blood flow (recirculation and flow jet formation observed) and pulse wave dynamics (derived pulse wave velocity = 4.67 $\pm$ 0.71 m/s). Integrative imaging of wall motion and blood flow in our phantoms also revealed fluid-structure interaction differences between healthy and diseased models. These findings show that phantoms developed with our new protocol are useful in vascular imaging studies that individually or jointly assess wall motion and flow dynamics.

9:20

2aBA4. Atherosclerosis characterization using lipid-specific photoacoustic imaging and 4D ultrasound strain mapping in mice. Gurneet S. Sangha and Craig J. Goergen (Biomedical Eng., Purdue Univ., 206 S. Martin Jischke Dr., West Lafayette, IN 47907, gsangha@purdue.edu)

Dual-modality photoacoustic tomography (PAT) and four-dimensional ultrasound (4DUS) imaging have recently been used study atherosclerosis progression in small animals. PAT uses pulsed laser light to induce acoustic waves and reconstruct lipid-specific compositional images of tissue. 4DUS captures dynamic volumetric information and can be used to estimate three-dimensional (3D) Green-Lagrange strain using a direct deformation estimation method. Here, we hypothesized that PAT/4DUS can be used to correlate changes in arterial strain and hemodynamics with lipid localization and density in animals that have undergone partial carotid ligation (PCL) induced-atherosclerosis. A 40 MHz transducer (Vevo2100, VisualSonics) and a ND:YAG pulsed laser (Surelite EX, Continuum) were used to image five apolipoprotein-E deficient mice that underwent PCL of the left carotid artery while being fed a Western diet. Animals were imaged using 4DUS at days 0, 1, 4, 7, 10, and 14 to obtain pulsed-wave Doppler for hemodynamic characterization and 4DUS images for strain mapping. At day 14 all animals were euthanized and 3D *in situ* PAT images of the left carotid artery were acquired using 1210 nm light. The results show that atherosclerotic lesions can be characterized via PAT to localize both lipid accumulation and density and 4DUS to identify regions of low strain.

9:35

2aBA5. Spatial analysis of cardiac strain using high-frequency four-dimensional ultrasound in mice. Frederick W. Damen, Arvin Soepriatna, and Craig J. Goergen (Biomedical Eng., Purdue Univ., 206 S. Martin Jischke Dr., Rm. 3083, West Lafayette, IN 47906, fdamen@purdue.edu)

Cardiac disease remains the number one cause for all mortality in the United States, prompting a continued effort to understand the various factors that exacerbate disease. In this endeavor, mouse models of cardiac disease have served a crucial role by allowing for both investigation of disease factors and longitudinal tracking of cardiac function. Routine assessment of cardiac function in mice can be acquired using high-frequency ultrasound; however, conventional techniques must rely on idealized cardiac geometries to calculate function metrics, as morphometric information is only available from a single plane. Aiming to overcome these limitations, our group has recently developed and validated a high frequency four-dimensional ultrasound (4DUS) technique that provides full volumetric information of the mouse heart synced over one representative cardiac cycle. Analysis of this 4DUS data can provide region-specific wall kinematic information, in contrast to global metrics such as ejection fraction and stroke volume. Preliminary applications of our technique have demonstrated abnormal left-ventricular contractile patterns in mouse models of cardiac hypertrophy, as well as ventricular remodeling in models of myocardial infarction. These initial efforts suggest that widespread adoption of 4DUS has the potential to help increase the amount of information obtained when using mouse models of cardiac disease.

9:50

2aBA6. Quantification of murine cardiac hypertrophy using 4D ultrasound. Alycia G. Berman, Jennifer L. Anderson, Elizabeth E. Niedert (Biomedical Eng., Purdue Univ., 206 S Martin Jischke Dr., MJIS 3083, West Lafayette, IN 47907, berman1@purdue.edu), Adrienne Scott, Corey P. Neu (Mech. Eng., Univ. of Colorado at Boulder, Boulder, CO), and Craig J. Goergen (Biomedical Eng., Purdue Univ., West Lafayette, IN)

Cardiac hypertrophy is abnormal thickening, followed by dilation, of the heart which can lead to congestive heart failure. Herein, we use a mouse model of hypertrophy to explore the relationship between *in vivo* strain and the resultant hypertrophic state. To do so, osmotic pumps containing saline (n = 5) or angiotensin II (AngII; n = 10) were surgically implanted into the dorsal flank of C57BL/6J mice. AngII increased blood pressure and cardiac afterload, causing myocardial hypertrophy. Mice were imaged weekly using a VisualSonics Vevo2100 ultrasound system with a MS550D transducer (40 MHz center frequency) to collect ECG-gated Kilohertz Visualization data. In combination with a linear stepper motor, we also collected four dimensional (4D) cardiac data (3D + time). Two weeks post-surgery, pumps were removed from a subset of mice to assess the heart's ability to repair itself post-insult (n = 5). All mice were euthanized at 4 weeks. Standard metrics of left ventricular mass measured via two-dimensional slices of the 4D data indicated significantly increased mass in the AngII mice by day 14. Removal of the pump enabled significant, but partial, recovery. Current work is being performed to calculate strain within the cardiac wall. Ultimately, we aim to determine if increases in *in vivo* strain precede increases in cardiac mass.

10:05–10:20 Break

Invited Papers

10:20

2aBA7. Advanced beamforming for improved functional assessment in echocardiography. Brett Byram, Kaz Dei, Siegfried Schlunk, and Adam Luchies (Biomedical Eng., Vanderbilt Univ., 2301 Vanderbilt Pl.; PMB 351631, Nashville, TN 37235, brett.c.byram@vanderbilt.edu)

Echocardiography is one of the most used medical imaging exams. The data from these exams are often used to compute quantitative metrics of cardiac health including measures such as the global longitudinal strain (GLS) or ejection fraction. Metrics related to blood flow are also derived from echocardiography data. These metrics have great potential because ideally, they provide quantitative biomarkers to monitor cardiac function over time and compare patient function to population values. Unfortunately, echocardiography data

is often severely corrupted by various forms of acoustic clutter including wavefront aberration, reverberation and off-axis scattering from bright structures like the ribs and lungs. These sources of clutter degrade ultrasound image quality and corrupt the ability to derive reliable quantitative biomarkers of cardiac function. To resolve problems like these, we have been developing non-linear (mathematically) beamformers. These include our aperture domain model image reconstruction (ADMIRE) beamformer and, more recently, our deep neural network beamformer. We have shown that these beamformers are able to reduce errors related to wavefront aberration, reverberation and off-axis scattering, and also that the ADMIRE beamformer can eliminate the underestimation of global longitudinal strain caused by high levels of stationary reverberation caused by the chest wall.

10:40

2aBA8. Super-resolution ultrasound imaging beyond the acoustic diffraction limit. Kang Kim (Medicine, Univ. of Pittsburgh, 950 Scaife Hall, 3550 Terrace St., Pittsburgh, PA 15261, kangkim@upmc.edu), Qiyang Chen, and Jaesok Yu (BioEng., Univ. of Pittsburgh, Pittsburgh, PA)

Contrast enhanced ultrasound (CEU) imaging technologies using microbubbles (MBs) provide superior contrast of vasculatures, effectively suppressing the surrounding tissue signals, but the spatial resolution remains to the acoustic diffraction limit. By localizing the center of each MBs unprecedented high spatial resolution beyond the acoustic diffraction limit can be achieved. However, some methods of localizing each center of the signals from individual MBs that only applies symmetrically distributed signal amplitude require a large number of imaging frames, especially when MBs are densely clumped, therefore result in a long scan time that is not ideal for *in vivo* scan under physiologic conditions. In this paper, we present an innovative approach using deconvolution technique that will allow for identifying signals from individual MBs from dense population in any forms even grouped together within the full-width-at-half-maximum (FWHM) of the point spread function (PSF) of the US probe. In this way, no collected frame sets require to be excluded for image reconstruction, therefore scan time can be reduced significantly. *In vivo* application of this new approach in identifying vasa vasorum in rabbit atherosclerotic plaque model will be presented. Some technical limitations including background noise as well as motion artifact will be discussed.

Contributed Papers

11:00

2aBA9. Motion-resistant vascular ultrasound imaging based on real-time eigen-filtering. Adrian J. Chee, Billy Y. Yiu, and Alfred C. Yu (Univ. of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada, alfred.yu@uwaterloo.ca)

Doppler flow imaging has become a standard clinical modality for vascular diagnostics. Nevertheless, it remains challenging to perform vascular ultrasound in more complicated diagnostic scenarios, because significant flashing artifacts may appear due to inadequate suppression of Doppler clutter arising from moving tissues. For one decade, we have been striving to achieve motion-resistant vascular ultrasound by designing advanced eigen-filtering algorithms whose attenuation response is adapted to clutter characteristics. Using a receiver operating characteristics analysis approach, we showed that in the presence of vessel pulsation and tissue vibration, our eigen-based motion-resistant signal processing chain yielded a significantly higher true positive rate (>90%) in depicting flow in comparison to non-adaptive signal processing chains. Another engineering challenge that we have overcome is the high computational demand of eigen-processing algorithms. We have successfully devised real-time implementations of eigen-based motion-resistant signal processing through designing parallel computing kernels that are executed on a graphical processing unit (GTX Titan X). In particular, we achieved real-time video-range throughput for full-view Doppler frames, up to a scan depth of 5 cm for slow-time ensemble length of 16 samples (i.e., beyond the typical requirement for carotid scans). These findings serve well to substantiate the practical feasibility of performing motion-resistant vascular ultrasound.

11:15

2aBA10. Ascertaining the relationship between acoustic droplet vaporization, inertial cavitation, and hemolysis. Newsha Jahanpanah (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, jahanpna@mail.uc.edu), Sneha Sharma, Karla P. Mercado-Shekhar, Haili Su (Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH), Hunter A. Palcich, Austin M. Wanek (Dept. of Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Kevin J. Haworth (Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH)

Mechanical stress associated with inertial cavitation (IC) can cause hemolysis. The goal of this study was to determine whether hemolysis occurs during acoustic droplet vaporization (ADV) and subsequent microbubble activity. Perfluoropentane droplets were manufactured using an amalgamation protocol, diluted 30-fold in porcine whole blood, and pumped through a flow phantom maintained at 37 °C. Droplets were insonified with a 5 MHz focused transducer (peak negative pressures of 0.25 MPa to 4.25 MPa, pulse length of 40 cycles, and a pulse repetition frequency of 500 Hz). A 128-element array was used to monitor the nucleation of ADV and to record cavitation emissions. Free hemoglobin in the effluents was assayed to detect hemolytic activity. Hemolysis, IC, and ADV were detected at pressures of 2.25 MPa and higher. At 4.25 MPa, 0.38 ± 0.08% hemoglobin was released post-hemolysis. A positive linear relationship was observed between the percent hemoglobin released and IC dose. Due to the concurrence of ADV and IC, it was not possible to determine whether ADV causes hemolysis independent of IC. These findings should be considered when planning studies of biomedical application of ADV. [Work supported in part by an Acoustic Society of America Robert W. Young Award.]

11:30

2aBA11. Frequency dependence of the vaporization threshold of sonosensitive perfluorocarbon droplets varying their liquid core and size. Mitra Aliabouzar (George Washington Univ., 8222 Harvest Bend Ln. APT. #37, Laurel, MD 20707, mitraaali@email.gwu.edu), Krishna Kumar (George Washington Univ., Washington, DC), and Kausik Sarkar (George Washington Univ., Washington, DC)

Phase shift liquid perfluorocarbon (PFC) droplets vaporizable by ultrasound have received increasing attention for therapeutic and diagnostic applications. The ultrasound activation pressure required for the phase change of these droplets into echogenic microbubbles is termed acoustic droplet vaporization (ADV). This study systematically investigates the effect of frequency of excitation¾2.25 MHz, 10 MHz and 15 MHz¾on ADV and inertial cavitation (IC) thresholds of a suspension of PFC droplets by varying the physical properties of the liquid droplets (size and boiling point) in a tubeless setup. We prepared lipid-coated droplets with three different liquid cores, perfluoropentane (PFP), perfluorohexane (PFH) and perfluorooctyle bromide (PFOB), of two different size ranges ¾ one with diameter smaller than 3 öŸm and the other with diameter larger than 10 öŸm. We found that the ADV threshold increases with frequency for the most volatile PFC liquid, PFP, for both large and small size droplets. While for the small size droplets of higher boiling point liquids, PFH and PFOB, no ADV was detected, for the larger ones ADV threshold decreases with frequency of excitation. ADV thresholds at all the frequencies studied here

occurred at lower rarefactional pressures than IC thresholds indicating that phase transition precedes inertial cavitation.

11:45

2aBA12. Acoustic droplet vaporization with microfluidic droplets results in dissolved oxygen scavenging. Rachel P. Benton (Neurosci. Baccalaureate Program, Univ. of Cincinnati, 213 River Valley Blvd., Cincinnati, OH 45157, bentonrp@mail.uc.edu), Newsha Jahanpanah (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, Cincinnati, OH), Emilee Warner (Dept. of Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Rohan S. Srivastava (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, Cincinnati, OH), Ishan Anand (Dept. of Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Kevin J. Haworth (Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH)

Acoustic droplet vaporization (ADV) is the ultrasound-mediated phase transitioning of liquid perfluoropentane (PFP) droplets into gas microbubbles resulting in dissolved oxygen scavenging from the surrounding fluid. The objective of this study was to determine how droplet diameter influences oxygen scavenging. Droplets of 12 μm and 6 μm diameters were manufactured using a microfluidic system, which were diluted in saline to obtain concentrations of 0.48 X 10^{-3} ð 0.12 X 10^{-3} ml/ml and 1.11 X 10^{-3} ð 0.13 X 10^{-3}ml/ml, respectively. Samples were pumped through a 37 °C flow phantom at 10 ml/min. A 5 MHz transducer insonified droplets at 5 MPa for 20 cycles with a 500 Hz repetition frequency. Oxygen partial pressure (Po_2) was measured with a distal sensor. Samples of 12 μm and 6 μm droplets had an average volume-weighted transition efficiency of 5.8% ± 4.5% and 5.3% ± 4.6%, respectively. The initial Po_2 was 169 ± 7 mmHg for all samples. ADV with 12 μm and 6 μm droplets reduced the Po_2 to 113 ± 21 mmHg and 101 ± 9 mmHg, respectively. An oxygen scavenging model based on the PFP phase transitioned volume yielded a final Po_2 of 121 ± 27 mmHg for 12 μm droplets and 106 ± 29 mmHg for 6 μm droplets. There was no statistically significant difference ($p > 0.05$) between the experimentally measured and modeled Po_2 values after ADV.

TUESDAY MORNING, 14 MAY 2019 EXHIBIT HALL, 8:00 A.M. TO 11:30 A.M.

Session 2aID

Interdisciplinary and Student Council: Graduate Programs in Acoustics (Poster Session)

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

Michael Rollins, Cochair
Biomedical Engineering, University of Cincinnati, MSB 6303, 231 Albert Sabin Way, Cincinnati, OH 45267

Matthew C. Zeh, Cochair
Mechanical Engineering Graduate Acoustics Program, University of Texas at Austin, 3374 Lake Austin Blvd., Apt. B, Austin, TX 78703

All posters will be on display from 8:00 a.m. to 11:30 a.m. To give contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:45 a.m. and contributors of even-numbered papers will be at their posters from 9:45 a.m. to 11:30 a.m.

Invited Papers

2aID1. Graduate acoustics at Brigham Young University. Tracianne B. Neilsen, Scott D. Sommerfeldt, Timothy W. Leishman, Kent L. Gee, Brian E. Anderson, Jonathan Blotter, Scott L. Thomson, and William Strong (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

Graduate studies in acoustics at Brigham Young University prepare students for industry, research, and academia by complementing in-depth coursework with publishable research. Graduate-level coursework provides students with a solid foundation in core acoustics principles and practices and measurement skills, including a strong foundation in experimental techniques and writing technical memoranda. Labs across the curriculum cover calibration, directivity, scattering, absorption, laser Doppler vibrometry, lumped-element mechanical systems, equivalent circuit modeling, arrays, filters, room acoustics, active noise control, and near-field acoustical holography. Recent thesis and dissertation topics include active noise control, directivity of acoustic sources, room acoustics, radiation and directivity of musical instruments, energy-based acoustics, time reversal, nondestructive evaluation, flow-based acoustics, voice production, aeroacoustics, sound propagation modeling, nonlinear propagation, high-amplitude noise analyses, machine and deep learning applied to ambient noise level prediction, crowd noise interpretation, underwater acoustic source localization, and ocean environment classification. The graduate students are expected to present research at professional meetings and publish in peer-reviewed acoustics journals. Graduate students often serve as peer mentors to undergraduate students on related projects and may participate in field experiments to gain additional experience. For update, follow us @BYUAcoustics

2aID2. Graduate programs in acoustics at the University at Buffalo, State University of New York. Anastasiya Kobrina and Kali Burke (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

University at Buffalo is known for its diversity in acoustic and auditory research spanning from signal processing to hearing in humans and animals. These research programs are nested within Anthropology, Biological Sciences, Communicative Disorders and Sciences, Computer Science and Engineering, Linguistics, Psychology, and Music, which leads to a unique variety of collaborations spanning departments and laboratories. The doctoral programs in these departments are aimed at training students for clinical and research-oriented careers. Our graduate programs involve taking extensive coursework and hands on experience in laboratories. In addition, each department holds regular colloquia on various topics in acoustics. University at Buffalo is also home to the Center for Hearing and Deafness. The Center seeks to develop cooperative working relationships with businesses and industries involved in hearing-related activities, such as: hosting the WNY Tinnitus Support Group, testing and evaluating drugs used to treat hearing loss, developing new scientific and clinical instrumentation, and assessing industrial hearing loss and noise regulations. The Center also provides valuable training opportunities for physicians, engineers, and health professionals. In conclusion, the University at Buffalo is an ideal fit for training in acoustics research.

2aID3. Biomedical acoustics research at the Image-Guided Ultrasound Therapeutics Laboratories. Christy K. Holland, T. Douglas Mast, and Kevin J. Haworth (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng., Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

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 laboratories, directed by Christy K. Holland, comprise graduate and undergraduate students, postdoctoral fellows, physician-scientists, and clinical and scientific collaborators in fields including cardiology, neurosurgery, neurology, radiology, and emergency medicine. 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. Imaging algorithms incorporate both passive and active cavitation detection. The Biomedical Acoustics Laboratory within IgUTL, directed by T. Douglas Mast, investigates ultrasound imaging for therapy guidance, including echo decorrelation imaging for monitoring liver cancer ablation and real-time tracking of tongue motion for biofeedback in speech therapy. The Biomedical Ultrasonics and Cavitation Laboratory within IgUTL, directed by Kevin J. Haworth, employs ultrasound-mediated gas scavenging for image-guided treatment of cardiovascular disease, especially reperfusion injury. [Work supported by NIH Grants R01 NS047603, R01 HL135092, R01 HL133334, R01 CA158439, R01 DC017301, and K25 HL133452.]

2aID4. Graduate studies in acoustics and wave physics at Institut d'Acoustique—Graduate School, Le Mans, France. Vincent Tournat (LAUM, CNRS UMR 6613, Le Mans Université, Av. O. Messiaen, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr)

This poster presents the graduate studies in Acoustics at Le Mans (France) offered at the Institut d'Acoustique—Graduate School. Graduate studies in Acoustics at Le Mans University have been awarded in 2018 the excellence label École Universitaire de Recherche among 28 other reference centers for all fields of Science, through a highly selective national call. Master and engineering school programs range from physical acoustics, environmental acoustics, acoustics and vibrations to international masters on electro-acoustics and on wave physics. The education through research is carried out at the LAUM, UMR CNRS, one of the largest acoustics laboratory in the world. Several details, objectives and contact informations on the graduates studies will be given on the poster.

2aID5. Graduate training opportunities in the hearing sciences at the University of Louisville. Shae D. Morgan (Dept. of Otolaryngol. and Communicative Disord., Univ. of Louisville, 627 S. Preston St., Ste. 220, Louisville, KY 40292, shae.morgan@louisville.edu), Hammam AlMakadma (Dept. of Otolaryngol. and Communicative Disord., Univ. of Louisville, Syracuse, New York), Maria V. Kondaurova, Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), and Pavel Zahorik (Dept. of Otolaryngol. and Communicative Disord., Heuser Hearing Inst. and 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.

2aID6. Graduate programs at the University of Maryland. Eric C. Hoover, Samira B. Anderson, Matthew Goupell, and Sandra Gordon-Salant (Dept. of Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, ehoover@umd.edu)

The University of Maryland has many opportunities for graduate studies in topics in acoustics, including hearing, speech and language sciences, neuroscience, physiology, bioacoustics, linguistics, and acoustical and communications engineering. Graduate training at the University of Maryland is characterized by an emphasis on multidisciplinary, collaborative research designed to enable students to develop a breadth of knowledge in addition to their focused research program. There are numerous interdisciplinary initiatives that address problems of great importance to science and society. In addition to the interdisciplinary research training on campus, students also train and collaborate with world-class researchers from nearby institutions including National Institutes of Health, Walter Reed National Military Medical Center, the University of Maryland Medical Center, and many other outstanding institutions in the area. Individuals interested in graduate training at the University of Maryland should contact faculty in their field of interest to learn more about the many opportunities available.

2aID7. Graduate studies in Acoustical Oceanography in the Massachusetts Institute of Technology and Woods Hole Oceanographic Institution Joint Program. Andone C. Lavery (Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

An overview of graduate studies in Acoustical Oceanography 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.

2aID8. Graduate research opportunities in acoustics at the University of Michigan, Ann Arbor. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

The University of Michigan (UM) is host to a wide array of acoustics research which encompasses many of the Technical Committees of the ASA. Within the Department of Mechanical Engineering alone, work is being done to develop better remote sensing techniques for underwater environments, to better understand the mechanics of the human cochlea, and to build metamaterials allowing for new and exotic acoustic behaviors. Within the UM Medical School, faculty and graduate students are constantly advancing techniques for diagnostic and therapeutic ultrasound procedures. In the Department of Naval Architecture and Marine Engineering, computational and experimental tools are being developed to enable better ship design. In Electrical Engineering, advances in MEMS fabrication yield even smaller and more efficient transducers for smarter devices. And researchers in the Linguistics Department are using the fundamental acoustic processes of speech to learn how humans effectively communicate. And while these are only a sample of the projects taking place at Michigan, new opportunities for acoustics research and collaboration open up each semester. Combined with a rich course catalogue, first-rate facilities, and great prospects for publication, these opportunities prepare UM graduate students for careers in industry and academia alike. Go Blue!

2aID9. Graduate study in physical acoustics at the University of Mississippi. Joel Mobley, Josh R. Gladden, Cecille Labuda (Phys. and Astronomy, Univ. of Mississippi, P.O. Box 1848, 1034 NCPA, University, MS 38677, jmobley@olemiss.edu), and Likun Zhang (Phys. and Astronomy, Univ. of Mississippi, Oxford, MS)

The University of Mississippi is a Ph.D. granting institution with an R1 Carnegie designation placing it among schools with the highest level of research activity. The Department of Physics and Astronomy at Ole Miss has a diverse range of research opportunities, including two groups associated with recent Nobel Prizes. Along with programs in Computational Physics, High Energy Physics, Atmospheric Physics and Gravitation, the department is affiliated with the National Center for Physical Acoustics (NCPA). NCPA is an 85 000 square foot standalone facility on the campus of the University of Mississippi dedicated to the physics and engineering applications of acoustics. It has research groups dedicated to ultrasound, infrasound, aeroacoustics, atmospheric propagation, porous media, and ocean acoustics. Graduate students in physics and engineering are pursuing Ph.D. and M.S. degrees at NCPA, and four faculty members from the Physics department have their research laboratories in the facility. In addition to NCPA, the Physics department has affiliations with the Laser Interferometer Gravitational Wave Observatory (LIGO), The European Center for Particle Physics (CERN), Fermilab and Belle II.

2aID10. Opportunities for graduate studies in acoustics within the College of Engineering at the University of Nebraska–Lincoln. Lily M. Wang, Erica E. Ryherd (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu), Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska - Lincoln, Lincoln, NE), and Jinying Zhu (Civil Eng., Univ. of Nebraska - Lincoln, Omaha, NE)

A number of faculty in the College of Engineering at the University of Nebraska–Lincoln (UNL) conduct research in acoustics and mentor graduate students. Within the Durham School of Architectural Engineering and Construction, based at UNL's Scott Campus in Omaha, Lily Wang and Erica Ryherd are active in architectural acoustics and noise (http://nebraskaacousticsgroup.org). In Civil Engineering, Jinying Zhu (also based on UNL's Scott Campus in Omaha) focuses in structural acoustics, using ultrasonic waves for concrete evaluation. In Mechanical and Materials Engineering, Joseph Turner (based at UNL's City Campus in Lincoln) studies ultrasound propagation through complex media for quantitative characterization of materials/microstructure (http://quisp.unl.edu). UNL is additionally home to an active student chapter of the Acoustical Society of America, the first to be founded in 2004. This poster will summarize the graduate-level acoustics courses and lab facilities at UNL within the College of Engineering, as well as the research interests and achievements of our faculty, graduates, and students.

2aID11. Graduate studies in acoustics at the University of New Hampshire. Daniel R. Howard, Anthony P. Lyons (Univ. of New Hampshire, Durham, NH), Jennifer L. Miksis-Olds (Univ. of New Hampshire, Durham, North Carolina), and Thomas C. Weber (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

The University of New Hampshire (UNH) offers several opportunities for graduate students interested in studying acoustics and its application. Faculty mentors who are expert in acoustic methods and technologies reside in a range of programs and departments that are largely focused on the use of acoustics in the marine environment, including biological science, earth science, mechanical engineering, natural resources and earth systems, ocean engineering, and oceanography. UNH faculty mentors who specialize in acoustics are active in the Animal Bioacoustics, Acoustical Oceanography, and Underwater Acoustics technical committees. Recent studies by faculty and students focusing on fundamental acoustic problems, such as those that would cause a graduate student to be a regular attendee of meetings of the Acoustical Society of America, have come largely from mechanical engineering, ocean engineering, biological sciences, and the newly formed School of Marine Sciences and Ocean Engineering. Graduate students in these programs of study have the opportunity for formal classroom training in the fundamentals of acoustics, vibrations, and advanced topics in acoustics as they pursue their graduate training.

2aID12. Graduate acoustics research at the University of New Haven. Eric A. Dieckman (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, edieckman@newhaven.edu)

The University of New Haven offers graduate study opportunities in acoustics, primarily through a Master of Science in Mechanical Engineering degree. This flexible program allows students to tailor a curriculum toward their interests in preparation for further graduate studies or entry into the workforce. Courses offered include Fundamentals of Acoustics, System Vibrations, Nondestructive Evaluation, and Wave Propagation and Scattering. All classes have a focus on hands-on projects. A one-semester project or two-semester thesis allows students structured time to work on research problems in acoustics.

2aID13. Pursuing graduate degrees in acoustics at Penn State. Victor Sparrow and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

The Graduate Program in Acoustics at Penn State is the only program in the U.S. 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 Graduate Program in Acoustics is administratively aligned with the College of Engineering and closely affiliates with the Applied Research Laboratory. Research areas include: structural acoustics, nonlinear acoustics, architectural acoustics, signal processing, aeroacoustics, biomedical ultrasound, transducers, computational acoustics, noise and vibration control, psychoacoustics, and underwater acoustics. Course offerings include fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, experimental techniques, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, biomedical ultrasound, flow induced noise, spatial sound and three-dimensional audio, acoustics of musical instruments. More than 700 Penn State Acoustics graduates serve widely throughout military and government labs, academic institutions, consulting firms and industry. This poster describes faculty research areas, laboratory facilities, student demographics, successful graduates, and recent enrollment and employment trends.

2aID14. A master's degree in acoustics through distance education from Penn State. Daniel A. Russell and Victor Sparrow (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

The Graduate Program in Acoustics at Penn State provides online access to graduate level courses leading to the M.Eng. degree in Acoustics. Lectures are broadcast live via Adobe Connect to students scattered around the world, while archived recordings allow working students to access lectures at their convenience. Students earn the M.Eng. in Acoustics degree by completing 30 credits of coursework (six required courses and four electives) and writing a capstone paper. Since 1987, more than 150 distance education students have completed the M.Eng. in Acoustics degree. Many other students take individual courses as non-degree students. Courses offered online include: elements of acoustics and vibration, elements of waves in fluids, 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, biomedical ultrasound, flow induced noise, spatial sound and 3D audio, acoustics of musical instruments. This poster describes the distance education experience leading to the M.Eng. degree in Acoustics from Penn State and showcases student demographics, capstone paper topics, enrollment statistics and trends, and the success of our graduates.

2aID15. Underwater acoustics and ocean engineering at the University of Rhode Island. Lora J. Van Uffelen, James H Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu)

Underwater acoustics is one of the primary areas of emphasis in the Ocean Engineering Department at the University of Rhode Island, the first Ocean Engineering program in the United States. The program offers Bachelors, Masters (thesis and non-thesis options) and Ph.D. degrees in Ocean Engineering. These programs are based at the Narragansett Bay campus, providing access to a living laboratory for student learning. Some key facilities of the program are an acoustics tank, a 100-foot-long wave tank, and currently the R/V Endeavor, a UNOLS oceanographic research vessel operated by the University of Rhode Island. A new Regional Class vessel is anticipated in 2021. At the graduate level, students are actively involved in research focused in areas such as acoustical oceanography, propagation modeling, acoustic positioning and navigation, geoacoustic inversion, marine mammal acoustics, ocean acoustic instrumentation, and transducers. An overview of classroom learning and ongoing research will be provided, along with information regarding the requirements of entry into the program.

2aID16. Graduate acoustics education in the Cockrell School of Engineering at The University of Texas at Austin. Michael R. Haberman, Mark F. Hamilton, Preston S. Wilson (Walker Dept. Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu), Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

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 the Cockrell School of Engineering. The core of this program resides in the Walker Department of Mechanical Engineering (ME) and the Department of Electrical and Computer Engineering (ECE). Acoustics faculty in each department supervise graduate students in both departments. One undergraduate and eight 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 five graduate acoustics courses, taught in alternate years, are on nonlinear acoustics, underwater acoustics, ultrasonics, architectural acoustics, and wave phenomena. 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.

2aID17. Graduate education and research in architectural acoustics at Rensselaer Polytechnic Institute. Ning Xiang, Jonas Braasch, and Todd Brooks (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu)

The Graduate Program in Architectural Acoustics has been constantly advanced from its inception in 1998 with an ambitious mission of educating future experts and leaders in architectural acoustics, due to the rapid pace of change in the fields of architectural-, physical-, and psycho-acoustics. Recent years the program's pedagogy using science, technology, engineering, and mathematics methods (STEM) has been proven to be effective and productive, including intensive, integrative hands-on experimental components that integrate architectural acoustics theory and practice. The graduate program has recruited graduate students from a variety of disciplines including individuals with B.S., B.Arch., or B.A. degrees in Engineering, Physics, Mathematics, Architecture, Electronic Media, Sound Recording, Music and related fields. Under this pedagogy and research environment graduate students have been succeed in the rapidly changing field. RPI's Graduate Program in Architectural Acoustics has since graduated more than 150 graduates with both M.S. and Ph.D. degrees. Under the guidance of the faculty members they have also actively contributed to the program's research in architectural acoustics, noise control, psychoaoustics, signal processing in acoustics as well as our scientific exploration at the intersection of cutting edge research and traditional architecture / music culture. This paper illuminates the evolution and growth of the graduate program.

2aID18. Opportunities for graduate studies in physical, engineering, and underwater acoustics at the Naval Postgraduate School. Oleg A. Godin, Kevin B. Smith, and Anne T. Pickens (Phys. Dept., Naval Postgrad. School, 33 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu)

The Departments of Physics and of Electrical and Computer Engineering at the Naval Postgraduate School offer programs of graduate studies in acoustics leading to MS degrees in physics, applied physics, and engineering acoustics and PhD degrees in engineering acoustics and applied physics. The program is interdisciplinary, with courses and laboratory work drawn principally from the fields of physics and electrical engineering. Subjects covered include waves and oscillations; fundamentals of physical and structural acoustics; the generation, propagation and reception of sound in the ocean; civilian and military applications of sonar systems; and acoustic signal processing. Topics of recent theses and dissertations include development and field testing of novel sensors for atmospheric and ocean acoustics, modeling and measurements of ambient noise and sound propagation in the ocean, sound scattering in underwater waveguides, acoustic vector sensors and vector field properties, acoustic communications, geo-acoustic inversion, acoustic remote sensing of the ocean, and acoustics of autonomous underwater and aerial vehicles.

TUESDAY MORNING, 14 MAY 2019 BREATHITT, 9:00 A.M. TO 11:45 A.M.

Session 2aMU

Musical Acoustics: General Topics in Musical Acoustics

Randy Worland, Chair
Physics, University of Puget Sound, 1500 N. Warner, Tacoma, WA 98416

Contributed Papers

9:00

2aMU1. Non-destructive correlation of Nigerian drum beat-pattern and pitch to detect a ripen watermelon. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com) and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

In the application of musical acoustics and speech sound, almost any type of Nigerian drum is used for communication. The agreeable successions of tones unlimited to interesting beat-patterns, pitch and rhythms uses shifted accents, non-accented rhythms, syncopations etc. On the daily basis of a watermelon harvesting and trade point, musical acoustics and speech sounds are applied passively in detecting and determining a ripen watermelon; the application of the Nigerian drum beat-pattern, pitch and intonation is an efficient procedure for ripeness detection of watermelon. Depending on how the pitches are lowered or accented, the melon ripeness is detected. The pitch-pattern analysis can be used to measure, determine and correlate the internal ripeness and quality of watermelon with pitch from a Nigerian drum. This method allows identification at a 60.0 % level of efficiency. Hence, the proposed method can reliably detect watermelon ripeness.

9:15

2aMU2. The effect of axial vibrations on the input impedance of the trumpet. Brooke Rodgers and Thomas Moore (Dept. of Phys., Rollins College, Box 2743, Winter Park, FL 32789, brodgers@rollins.edu)

It has been shown that the vibrations in the bell section of the trumpet can influence the sound produced by the instrument. Recent research has indicated that these effects may be traced the axial vibrations of the bell. We report experimental results that relate the input impedance of the trumpet to the phase of externally induced longitudinal motion. These results indicate that the phase difference between the driver and the bell motion can significantly affect the input impedance. [Work supported by NSF Grant #PHY-160749.]

9:30

2aMU3. Nonlinear generation of sum frequencies in Sitka spruce. Jade Case, Lauren Neldner, and Thomas Moore (Dept. of Phys., Rollins College, Box 2743, Winter Park, FL 32789, jcase@rollins.edu)

It has been known for several decades that there are anomalous frequency components in the sound of a piano. These occur at the sum and difference frequencies of the overtones attributable to string motion, and are commonly referred to as phantom partials. Recent work has shown that, contrary to the generally accepted theory, the majority of power in these frequency components originates in the wooden parts of the piano and not the string. However, the etiology of the nonlinearity is not clear. To determine if the source of the nonlinearity is the stress caused by the pressure of the strings on the soundboard, experiments were performed on isolated pieces of Sitka spruce. The results indicate that stress can significantly increase the nonlinear response of the wood. [Work supported by NSF Grant #PHY-160749.]

9:45

2aMU4. High frequency cavity modes of a Helmholtz resonator excited by an air jet. Emma Shaw (Phys., Agnes Scott College, 141 E. College Ave., Decatur, GA 30030, eshaw@agnesscott.edu), James P. Cottingham, and Robert Stills (Phys., Coe College, Cedar Rapids, IA)

In 1990, Khosropour and Millet reported on a study of the internal spectrum of a Helmholtz resonator excited by an air jet and reported that the data for frequency and amplitude of the Helmholtz mode as a function of jet speed show a series of domains separated by narrow transition regions [*JASA* **88**, 1211–1221 (1990)]. The idea was to observe the behavior of the air jet when the resonator effectively has just a single mode, and does not overblow to higher frequency modes at moderate jet velocities. Further research explored the effects of changes in neck length and jet angle, but higher resonator modes were ignored. This paper presents characterizations of some of the higher modes of cavity vibration, which can be excited at high jet velocities. Unlike the Helmholtz mode, these modes depend on the shape of the cavity shape as well as the volume. Several of these modes were observed, sometimes with more than one mode sounding simultaneously. Guided by finite element simulations, the nature of these higher cavity modes was verified experimentally by observing frequencies and locating nodal surfaces.

10:00–10:15 Break

10:15

2aMU5. Comparative study of oboe and clarinet. Laura Fitzgerald and Gordon P. Ramsey (Loyola Univ. Chicago, 1032 W Sheridan Rd., Chicago, IL 60660, lfitzgerald1@luc.edu)

The oboe and clarinet are relatively similar instruments in their size and shape yet create such different sounds. This study is an exploration of the reasoning behind their differences through a detailed acoustical analysis and geometrical study of these instruments, comparing and contrasting their properties. The oboe and clarinet are comparable in size but have some key differences. The oboe's conical shape allows for all the harmonics to be present, while the clarinet is mostly cylindrical, except for the bell, so that mostly odd harmonics are present. The oboe's keys have holes much smaller than those of the clarinet. Looking to find differences in acoustical spectra based on their geometric differences, samples of low, medium, and high ranges on both instruments using the same concert pitches have been taken. Data have been taken in a regular lab setting as well as an anechoic chamber in order to look for differences in harmonic content for each instrument. The data are consistent with the notion that the shape of the instruments does contribute to the spectra.

10:30

2aMU6. Control of vocal loudness in singing. Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@utah.edu)

Vocal loudness in sones is quantified on the basis of changes in spectral slope and harmonic tuning in a range of singing fundamental frequencies of 125 Hz to 1000 Hz. Spectral slope of the mouth output pressure is bracketed in the range of -3 dB/octave to -12 dB/octave to reflect a typical glottal spectrum from breathy to pressed adduction. To approximate formant tuning of harmonics, the SPL level of the first three harmonics is raised by 10, 20, and 30 dB. It is shown that spectral slope change is more effective in increasing vocal loudness than tuning a single harmonic with a vocal tract resonance. Some applications to amplified and unamplified vocal production and respective training are given.

10:45

2aMU7. Feedback control of acoustic musical instruments when the number of sensors and actuators differ. Edgar J. Berdahl (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

Actuated instruments can be created in a variety of ways. An interesting class of actuated instruments are feedback-controlled acoustic musical instruments. A robust way to create these is to employ pairs of collocated sensors and actuators, and then to use the feedback control to simulate virtual physical systems. In the linear and time-invariant case, this means that the feedback control functions can be designed to be positive real. Accordingly, the physical properties of the instrument can become adjustable via the feedback control. An interesting case arises when the number of actuators and sensors differ. The actuators and sensors can however still potentially be collocated, which will result in the individual transfer functions from the actuators to the sensors (e.g. the mobilities) in being positive real. Under some special cases, stable feedback control can still be attained for a wide variety of feedback gains. The Feedback Guitar serves as an interesting case study for this. It has one actuator, which is approximately collocated with six piezoelectric sensors, one for each string. Using any non-negative linear combination of the sensor signals, an approximately positive real mobility can be obtained, which can enable stable feedback control for a wide variety of feedback gains.

11:00

2aMU8. Metamaterials in musical acoustics: A modified frame drum. Rolf Bader (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de), Jost L. Fischer, Malte Münster, and Patrick Kontopidis (Inst. of Systematic Musicology, Univ. of Hamburg, Hamburg, Germany)

Acoustic metamaterials have properties not found in nature, like negative stiffness, cloaking behavior or frequency bandgap damping. This is achieved by complex geometries often constructed out of multiple subunits in sub-wavelength size. Although also musical instruments often have complex shapes, like guitar or piano soundboards with regular fan bracing, metamaterials have not explicitly been used here. As an example, a modified frame drum is proposed with increased sound possibilities by adding masses to the drum membrane arranged in a circle. Such structures have been shown to have cloaking behavior. Using microphone-array and laser interferometry measurements it is shown that such a drum has a frequency-dependent cloaking behavior. When struck at the center of the added circle most energy above about 400 Hz stays in the circle and decays strongly. Such a sound cannot be produced with a regular frame drum. When struck outside the circle the drum sounds very much like a regular drum without added masses. By gradually changing the playing position from the circle center towards the circle rim, frequencies above about 400 Hz are gradually added. Therefore such a modified frame drum has much more possible sounds and therefore ways of musical articulation.

11:15

2aMU9. Automatic transcription of solo audio into music notation. Dong Hyun Lee (Dept. of Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL 61801, dlee134@illinois.edu) and James W. Beauchamp (Dept. of Elec. and Comput. Eng., School of Music, Univ. of Illinois Urbana-Champaign, Urbana, IL)

Creating sheet music can be an arduous task for instrumentalists and vocalists who improvise. This problem can be solved by using a pitch detector that produces a MIDI data file. MIDI data is a good solution because it can be easily transformed into music notation by standard programs such as

Logic Pro X or Finale. The authors have developed an open source C/UNIX-based program that automatically transforms a monophonic sound file into a playable MIDI file. Pitch (F0) detection is accomplished using a short-time autocorrelation algorithm. Successive F0's that correspond to the same MIDI note number are combined to form notes. The minimum duration of each note is determined by the autocorrelation window size, which in our case is set to 0.03 s. To achieve a more accurate notation result, the program employs duration and RMS amplitude thresholds to exclude spurious notes from the MIDI data.

11:30

2aMU10. A Spatially Distributed Vibrotactile Actuator Array (SDVAA) for music-to-vibrotactile sensory augmentation. Edgar J. Berdahl, Austin Franklin (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu), and Eric Sheffield (Music, Louisiana State Univ., Atlanta, GA)

The design of a Spatially Distributed Vibrotactile Actuator Array (SDVAA) is presented. It employs a multitude of vibrotactile actuators in order to communicate a larger amount of information than is possible using a single actuator. The SDVAA is currently being used for applications in music-to-vibrotactile sensory augmentation. While prior related projects have focused more on sensory substitution, this project aims only to add to a person's experience by augmenting a multimedia presentation with vibrotactile feedback. Because haptic perception is fundamentally different than auditory perception, it makes sense to rearrange the information transmitted to the haptic senses. For example, while auditory perception is limited to approximately the range 20 Hz to 20 kHz, tactile perception is limited primarily to the range 0 Hz to 800 Hz (if not higher). Accordingly, one approach being considered for converting auditory signals to vibrotactile signals is pitch shifting. Project results relating to music composed specifically for the SDVAA as well as general musical vibrotactile prototyping concepts will be presented.

TUESDAY MORNING, 14 MAY 2019 SEGELL, 8:00 A.M. TO 11:15 A.M.

Session 2aNS

Noise, Architectural Acoustics, Structural Acoustics and Vibration, and ASA Committee on Standards: Structure-Borne Noise in Buildings and What We Can Do About It

Bonnie Schnitta, Cochair
SoundSense, LLC, 46 Newtown Lane, Suite One, East Hampton, NY 11937

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

Chair's Introduction—8:00

Invited Papers

8:05

2aNS1. Stucture-borne noise from a pool. Felicia Doggett (Metropolitan Acoust., LLC, 1628 JFK Blvd., Ste. 1902, Philadelphia, PA 19103, f.doggett@metro-acoustics.com)

A problem arose at a recently completed high-rise condominium building in Philadelphia. Tenants on many levels above a pool located on the second level of the building stressed that they could hear low-frequency sound when an early-morning swimmer was doing laps. Looking through literature, we found nothing to attest to the phenomenon that presented itself. To investigate, we used a 15 lb rubber medicine ball to simulate the cavitation caused by the swimmer, which also provided a repeatable source. Cavitation in the water created by the arm strokes of the swimmer was exciting the pool shell and transmitting through the structure ultimately converting to audible sound via the lightweight partitions in the living units. Our measurement system included time synchronized triaxial accelerometers to determine the transmission paths through the building, which identified two significant peaks at 35 Hz and 62 Hz. This case study details this unique problem, the identification of structural transmission, and the successful outcome after our recommendations were implemented.

8:25

2aNS2. Case study of acoustic disturbances in pencil towers due to building movement. Sean Harkin, Jennifer Scinto (Eng., SoundSense, LLC, P.O. Box 1360, Wainscott, NY 11975, sean@soundsense.com), and Bonnie Schnitta (Eng., SoundSense, LLC, East Hampton, NY)

Pencil structure high-rise residential towers are becoming increasingly prevalent in major city centers as the demand for housing increases in urban centers with limited available property for development. The narrow footprint of these structures, combined with the efforts to build subsequent structures higher than the last, lead to unique acoustic issues within these buildings, which impact the residents' quality of life and affect the value of the properties. During periods of high, and even medium wind speeds, these structures deflect more than traditional skyscraper buildings and the resulting moment force on the internal building assemblies creates disturbing acoustic events that have been measured to be significantly higher than ambient sound levels in select octave bands. This paper presents two case studies in one such pencil structure building in New York City, from diagnosis, to proposed and implemented innovative acoustic treatments, to client satisfaction.

8:45

2aNS3. Prevention of structural shorts in highly engineered noise and vibration isolation systems. Alexander C. Born (Getzner USA, Inc., 8720 Red Oak Blvd., Ste. 400, Charlotte, NC 28217, alexander.born@getzner.com) and Sean Harkin (SoundSense, LLC, Wainscott, NY)

Noise and vibration isolation in structures is becoming more and more common as the world we live in becomes noisier and noisier. The increase in noise coupled with people's desire for lower background noise and vibration criteria results in the need for higher performing isolation systems. In order to achieve these performance levels, the system needs to be properly designed and engineered. One of the most common culprits for a system not providing the expected performance calculated during the design process is a short or sound bridge . A short typically occurs during the installation of the system by rigidly connecting the structure or machine that is being isolated to an unisolated element. This causes noise and vibration from sources such as subway, transit, and mechanical equipment to transfer with ease into the structure intended for isolation, rendering the highly engineered system useless or highly degraded. This paper will examine a multitude of ways to eliminate the possibility of a short and elaborate on the importance of the installation process of isolated systems. It will also review how the process differs when isolating for vibration compared to noise and the importance of each.

9:05

2aNS4. Heavy-weight impact testing: Test repeatability for a modified kettlebell and comparison of impact source weights and drop heights. Michael Raley (Ecore Int., 715 Fountain Ave., Lancaster, PA 17601, mike.raley@ecoreintl.com)

Noise and vibration from heavy-weight impacts associated with fitness activities are a common source of complaints in hotels, multi-family residences, and corporate office spaces. *In situ* testing using various heavy/hard impact sources is commonly used to evaluate the effectiveness of mitigation measures to address noise and vibration complaints. Previous work (LoVerde *et al*., Internoise 2015) proposed the use of a 16-lb spherical shot for *in situ* testing. However, significantly heavier weights are common in many gyms. For instance, hotel chains commonly include free-weights with a range of 5 to 75 lb in their fitness rooms. It is not known if the relatively light 16-lb shot provides results that are indicative of impacts from significantly heavier weights. This presentation evaluates the use of heavier (50-lb and 100-lb) kettlebells and varying drop heights as alternates to the 16-lb shot. Additionally, this presentation evaluates test repeatability for the kettlebells which have been modified so that the impact surface is spherical in shape.

9:25

2aNS5. Too many Cooks in the kitchen?: A review of noise and vibration challenges in mixed-use buildings for the session: Structure-borne noise in buildings and what we can do about it. Sarah Taubitz (45dB Acoust., LLC, 45dB Acoust., LLC, PO Box 12275, Denver, CO 80212, st@45db.com)

Several projects with rooftop structural vibration and/or noise transmission through structures, and footfall impact issues will be shared, along with a review of recommendations and outcomes of the example projects. We plan to discuss noise/vibration challenges such as: (1) Design, or re-design, of Heating, Ventilation, and Air-conditioning (HVAC) and kitchen exhaust fan (KEF) on rooftops and within building envelopes. (2) Finding noise leaks in structure. (3) Is it the MRI machine's vibration, or the Variable Air Volume (VAV) box causing noise and vibration in the office upstairs? (4) Discussion of resonance of different floor/ceiling assemblies.

9:45–10:00 Break

10:00

2aNS6. Efficacy of high-performance ceiling systems. Wilson Byrick (Pliteq, 1370 Don Mills Rd., Unit 300, Toronto, ON M3B 3N7, Canada, wbyrick@pliteq.com) and Matthew V. Golden (Pliteq, North York, ON, Canada)

A current approach to try to solve a low frequency impact-induced structure-borne noise issue in lightweight woodframe buildings has been the application of a resilient ceiling suspended by one-inch deflection spring hangers. While this system has been lab tested it was only compared to an assembly with no isolation in which the gypsum board was directly attached to the bottom of the joists. The authors have repeated the testing of one-inch deflection ceiling hangers in the same lab but this time they compared it to the same isolators with the springs removed, standard resilient sound clips and other ceiling isolation systems. Relative performance differences of the isolation systems will be presented along with transmissibility, static stiffness, dynamic stiffness and damping measurements of each isolation element. A few theories as to why the systems behave the way they do will also be presented.

10:20

2aNS7. Measurements of rail vibration in a residential building behind a wave barrier trench. David W. Dong and John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

The authors have previously reported on a wave barrier trench that was constructed to reduce vibration transmission from railroad to a multifamily residential building [*J. Acoust. Soc. Am.* **139**, 2159]. The reduction in ground vibration levels due to the trench was measured in several site conditions [*J. Acoust. Soc. Am.* **140**, 3281]. This paper extends the study with additional measurements of vibration transmission into the building and structure-borne propagation within the building.

10:40

2aNS8. Developments in resonant, low frequency sound absorbing devices: trials, errors and successes. Jeffrey Madison (RPG Acoust. Systems, LLC, 99 South St., Passaic, NJ 07055, jmad008@hotmail.com)

Nowhere is it more prevalent in the acoustical treatment world where the predicted can deviate from the actual outcome than in resonant, low frequency sound absorber design and construction. Any worthwhile resonant sound absorber (RSA) is a system, multiple parts working together toward a common design goal. RSAs can be designed via mathematical prediction, Helmholtz much more accurately than membrane configurations. Regardless of the predictions, when the time comes to construct RSAs, each and every component that goes into and that must work with the other components in the system, whether in a factory setting or in field construction, will have a significant influence on the actual outcome. Prediction techniques swiftly identify resonant frequencies, for example, however can miss widely on bandwidth and efficiency due to variables introduced during construction. It is therefore important to use experimentation and measurement techniques to understand actual outcome possibilities. One example of this is utilizing a large format impedance tube that measures designs effectively down to 20 Hz. Examples of RSAs along with their development and testing will be presented.

Contributed Paper

11:00

2aNS9. Estimating the acoustic power of sources in semi-reverberant enclosures using generalized energy density. Travis N. Hoyt, Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., N306 ESC, Provo, UT 84602, travisnhoyt@gmail.com), and Jonathan Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Sound power measurements of acoustic sources are typically performed in anechoic or reverberation chambers using acoustic pressure according to international standards. The anechoic chamber creates a free-field environment where the sound power is estimated from the squared pressure integrated over some enveloping surface. The reverberation chamber produces diffuse-field conditions, where sound power is proportional to the spatially averaged squared pressure. Since most acoustic sources exist in rooms that are neither anechoic nor entirely reverberant, it is desirable to estimate the sound power within these non-ideal, semi-reverberant spaces. In such environments, the direct and reverberant energies each contribute to the total measured field. If the kinetic and potential components of acoustic energy density are weighted appropriately, the spatial variation of the field can be significantly reduced compared to squared pressure. This generalized energy density allows an adaptation of the sound power formulation by Hopkins and Stryker to be used to make an efficient and accurate *in situ* sound power estimate of a noise source in a non-ideal acoustical environment. Since generalized energy density optimizes the spatial uniformity of the field, fewer measurement positions are needed compared to traditional standards. The experimental results and practical limitations of this method will be discussed.

Session 2aPA

Physical Acoustics and Noise: Nonlinear Acoustics for Non-Specialists I

Won-Suk Ohm, Cochair
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Kent L. Gee, Cochair
Brigham Young Univ., N243 ESC, Provo, UT 84602

Chair's Introduction—7:55

Invited Papers

8:00

2aPA1. Early history of nonlinear acoustics: Waveform distortion, disaster, and redemption. David T. Blackstock and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, dtb@austin.utexas.edu)

The first (although slightly incorrect) wave equations for finite-amplitude sound in lossless fluids were obtained independently by Euler in 1759 and Lagrange in 1760. Poisson (1808) provided the first major breakthrough with his exact solution for progressive waves of finite amplitude in a lossless gas. Although a far-reaching result, the progressive waveform distortion (and disastrous consequences) implied by his solution went unrecognized for 40 years. Challis (1848) showed that the Poisson solution is not single valued but did not understand why. Stokes (1848) provided the why. He saw that the Poisson waveform distorts as the wave travels, eventually threatening to become multivalued. He postulated that a discontinuity (shock) develops to avoid waveform overturning. He also saw that viscosity (not accounted for by Poisson) would prevent true discontinuities. Earnshaw (1860) and Riemann (1860) cleaned up plane waves in lossless gases. However, how to predict propagation after shocks form? Rankine (1870) and Hugoniot (1887, 1889) provided the first solutions for propagation when dissipation is included. These were the first steps toward redemption. The curtain rang down on this era of nonlinear acoustics with the excellent papers in 1910 by Rayleigh and Taylor on steady shocks in a thermoviscous fluid.

8:20

2aPA2. Overturning of nonlinear compressional and shear waveforms subject to power-law attenuation or relaxation. John M. Cormack and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

Model equations for plane nonlinear wave propagation that do not account for energy loss inevitably predict waveforms that are multivalued. Referred to as waveform overturning, this reveals that essential physics is not represented in the mathematical model. Energy loss from thermoviscous absorption keeps waveform steepening in check and prevents overturning for all source amplitudes, thus ensuring physical relevance of solutions. This is not the case for every energy loss mechanism, and waveform overturning may still occur above a critical source amplitude despite the presence of losses. The present work determines the critical source amplitude associated with two loss operators, one for power-law attenuation and the other for relaxation, and for both compressional waves (quadratic nonlinearity) and shear waves (cubic nonlinearity) [Cormack and Hamilton, *Wave Motion* **85**, 18 (2019)]. Reformulation of the model equations in intrinsic coordinates enables numerical solution of waveform evolution up to and beyond the distance at which waveform overturning occurs. It is found that attenuation and dispersion resulting from a power law with exponent less than unity, or the attenuation and dispersion due to relaxation, is insufficient to prevent waveform overturning at all amplitudes. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

8:40

2aPA3. Jet crackle: From production to propagation to perception. Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

The sound of a crackling jet is unusual. It is an irregular, popping noise that gives a distinct impulsive quality to the broadband noise from military jet aircraft, rockets, and even volcanoes. What are its possible causes? Which factors influence its propagation—and it perception? This presentation reviews historical and recent research into a phenomenon that lies at the crossroads of fundamental physical and psychoacoustics.

9:00

2aPA4. Sonic booms and sonic thumps for non-specialists. Victor Sparrow (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

One of the most straightforward applications of nonlinear acoustics is in the long range propagation of the shock waves created by supersonic aircraft. The sounds heard on the ground are either loud (sonic booms) or are quiet (sonic thumps) depending on the pressure versus time signature. This talk with give a brief overview of sonic booms and the role nonlinear acoustics plays in their propagation. The carpet of sounds heard on the ground from an example single flight will be shown. The role of nonlinear acoustics in the certification of future supersonic low-boom aircraft will be highlighted. Much more information about sonic boom is freely available from ntrs.nasa.gov, and one great reference is "Sonic Boom" by Maglieri, *et al*. (NASA/SP-2014-622). [Work supported by the U.S. Federal Aviation Administration Office of Environment and Energy through ASCENT, the FAA Center of Excellence for Alternative Jet Fuels and the Environment, Project 41 through FAA Award Number 13-C-AJFE-PSU under the supervision of Sandy Liu. Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

9:20

2aPA5. Nonlinear acoustics at Moscow State University. Oleg A. Sapozhnikov and Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow 119991, Russian Federation, and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Washington, oa.sapozhnikov@physics.msu.ru)

The rapid development of nonlinear acoustics, which began in the second half of the 20th century, had several distinct centers in different countries, and one of them was M. V. Lomonosov Moscow State University (MSU) in Russia. From MSU came one of the basic equations in nonlinear acoustics—the Khokhlov-Zabolotskaya (KZ) equation, the 50th anniversary of which is celebrated this year [*Sov. Phys. Acoust.* **15**(1), 35–40 (1969)]. By the 1960–1970s, the work conducted by Academician Rem Khokhlov and his colleagues made MSU a leading international center for nonlinear acoustics and, even more influentially, for nonlinear optics. Nonlinear optics at that time underwent rapid growth related to the invention of lasers, and many phenomena in optics had analogs in acoustics. One of them was a parabolic approximation for wave beams, which led to formulation of the KZ equation after translation to acoustics with consideration of the details of nonlinear effects in weakly dispersive media. Along with the development of theoretical foundations, initial experiments on nonlinear acoustic phenomena were carried out at MSU. The current paper highlights these studies, as well as contemporary work and achievements on nonlinear acoustics at Moscow State University.

9:40

2aPA6. Audio spotlight: Sound from ultrasound. Joseph Pompei (Holosonics, 400 Pleasant St., Watertown, MA 02472, fjpompei@holosonics.com)

The Audio Spotlight® is the trade name of the first and only commercially successful ultrasonic loudspeaker, and delivers a laser-like beam of sound as tight as a beam of light. This allows audio to be delivered to one listener or a small group, without disturbing others nearby, or sound can be projected to create "true surround sound" without multiple loudspeakers. Originally developed at MIT, the technology is used widely in a variety of fields, such as museums, libraries, hospitals, and retail displays. This invited lecture will present an overview of the nonlinear acoustics as a basis for such a device, as well as some specific engineering challenges. Commercial applications will be discussed, and live demonstrations are included.

10:00–10:15 Break

Contributed Papers

10:15

2aPA7. Comparison of sonic boom propagation models with measurements above the Earth's turbulent boundary layer. Joel B. Lonzaga (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., B1208 MS 463, Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

NASA conducted the Sonic Booms in Atmospheric Turbulence (SonicBAT) overflight campaign at Armstrong Flight Research Center in 2016 and Kennedy Space Center in 2017. The sonic booms were generated by F-18 aircraft flying at nearly constant altitudes and at nearly constant, supersonic speeds. Although the primary objective of the campaign was to investigate the effects of turbulence on the propagation of sonic booms from the aircraft, a subset of the SonicBAT datasets can be used to validate sonic boom propagation models. This dataset is obtained from the recording using a microphone mounted at a wingtip of a TG-14 glider flying above the Earth's turbulent boundary layer. With this dataset, effects due to turbulence and ground impedance, which are still poorly understood, are effectively eliminated. Consequently, comparison is made between measurements and models which only account for the nonlinear and absorption effects. Such models are obtained using NASA's PCBoom sonic boom propagation code, which has recently been updated to incorporate the full wind effects and improve the code's accuracy and efficiency. Additionally, near-field signatures of the aircraft using PCBoom's near-field approximant are also compared with those generated by The Boeing Company using computational fluid dynamics.

10:30

2aPA8. Finite difference time domain investigation of interior sound fields generated by parametric acoustic arrays. Anpeksh A. Saksena and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., Columbus, OH 43210, saksena.13@osu.edu)

Parametric acoustic arrays enable tight beams of audible sound through the nonlinear collimation and mixing of ultrasonic acoustic waves. Previous investigations have considered parametric array characterization when operated in free field environments and in the presence of rigid planar reflectors, so that the opportunity to leverage parametric arrays in more realistic environments with diffusive scatterers and absorptive surfaces remains unexplored. To fill the knowledge gap, this research establishes a finite difference time domain (FDTD) model of a parametric array in an interior environment with a combination of reflective, absorptive, and diffusive bounding surfaces. The modeling framework builds upon existing FDTD models of nonlinear ultrasound wave propagation used in the life sciences and medicine. Here, the studies seek to elucidate how the sonic sound field

is manipulated when the reflective surfaces that receive the nonlinear ultrasonic waves are neither planar nor perfectly reflective. The relative ability to take advantage of such interaction between incident ultrasound and reflected energies, when compared to a perfectly rigid plane for reflection, is examined in detail. All together, this work establishes and harnesses an FDTD modeling framework to examine the use of parametric acoustic arrays in conventional interior environments having reflective, absorptive, and diffusive surfaces.

10:45

2aPA9. Nonlinear focusing of high amplitude sound using time-reversal. Brian D. Patchett, Brian E. Anderson, Matthew L. Willardson, and Michael Denison (Phys. & Astronomy, Brigham Young Univ., N203, Eyring Sci. Ctr., Provo, UT 84606, brian.d.patchett@gmail.com)

Time reversal (TR) is a signal processing technique that may be used to intentionally generate high amplitude focusing of sound. The use of time reversal in room acoustics has been studied by others, but the application to generating high amplitude focusing has not previously been explored. The purpose of this study is to generate high amplitude sound waves in order to mimic a virtual spherical source with enough intensity to observe nonlinear wave propagation. Experiments have been carried out in a reverb chamber with eight compression horn drivers. Using these drivers, the impulse response is calculated, reversed in time, and modified using the clipping technique. When these signals are broadcast from the sources, a focus is generated at the receiver location with peak levels reaching 198 dB (ref 20 μPa). As the waves superpose at the focus, the amplitudes observed do not scale linearly when the experiment is repeated at increasing amplification levels. The compression peaks are higher in amplitude than expected, and the rarefaction troughs are lower in amplitude (less negative) than expected, when compared to linear scaling. Additionally, the diverging waves from the focus resemble the propagation of a single spreading shockwave.

11:00

2aPA10. Comparison of two ultrasonic backscatter coefficient methods under nonlinear distortion. Andres Coila and Michael Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, acoila@illinois.edu)

The backscatter coefficient (BSC) is a fundamental property of tissues and can be parameterized for tissue characterization. The BSC requires a reference signal, which is estimated through either the planar reflector method (PRM) or the reference phantom method (RPM). In both methods, linear acoustic propagation is assumed. In this work, the BSC estimation methods are evaluated when nonlinear distortion is present. RF data were acquired from two physical phantoms, labeled A and B, with a 5 MHz single-element transducer using low power (1 excitation level) and high power (6 increasing excitation levels) excitation signals. Phantom A contained glass beads with diameters ranging from 75 to 90 μm and phantom B had glass beads with diameters ranging from 9 to 43 μm. The BSCs estimated using the low power setting and high power settings were compared for the both the PRM and RPM through a root mean square error (RMSE). Estimates of the effective scatterer diameter (ESD) were obtained using each method for each high power setting and compared to the low power setting. The RMSE increased as the power setting increased with much higher RMSEs using the PRM compared to RPM, i.e., a maximum of 7.3 times larger for phantom A and 8.6 times larger for phantom B. Estimates of ESD matched the ranges of glass beads sizes for both phantoms except when using the PRM at higher power settings. These findings suggest that the RPM is more robust to nonlinear distortion effects compare to the PRM.

11:15

2aPA11. Enhancing dynamic positioning performance inside mid-air acoustic levitator. Tatsuki Fushimi (Mech. Eng., Univ. of Bristol, Queen's Bldg., Bristol BS8 1TR, United Kingdom, t.fushimi@bristol.ac.uk), Asier Marzo (UpnaLab, Universidad Pública de Navarra, Bristol, United Kingdom), Thomas L. Hill, and Bruce W. Drinkwater (Mech. Eng., Univ. of Bristol, Bristol, United Kingdom)

Acoustic levitators are devices which generate converging acoustic radiation forces and thus can trap objects in mid-air. Acoustic levitators have found applications in the fields of chemistry and biology as a non-contact transportation method. The trapping of objects can be achieved using a phased-array in which the phase of the signal sent to each transducer is varied to generate a trap that stably holds the particle at the target three-dimensional position. The transducer phases can be changed over time to translate the acoustic field in space, thereby transporting the trapped particle. Here, we describe an open-loop spatial calibration scheme which increases the positioning accuracy of a particle in an acoustic levitator. The effectiveness and the performance of the spatial calibration was determined using a single-axis standing wave levitator with 60 ultrasonic transducers (40 kHz), and a levitated particle (EPS particle of radius 0.7 mm). Our calibration method is shown to significantly improve the positioning accuracy of the particle inside the acoustic levitator and reduced RMS error down to 0.11 and 0.03 mm in x and z axes, respectively. Although the calibration approach only considers the static response, the trajectory when the particle moves at a relatively high velocity ($\approx$1 cm/s) was also improved. Increasing the precision and velocity of a moving particle will enhance the capabilities and reliability of acoustic levitators and open up possibilities for novel applications.

11:30

2aPA12. Born approximation of acoustic radiation force and torque on soft objects of arbitrary shape. Thomas S. Jerome, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029, tsjerome@utexas.edu)

When the density and compressibility of an object are similar to the corresponding properties of the surrounding fluid, and the dominant contribution to the acoustic radiation force depends on gradients of the energy densities as occurs in plane standing waves, the Born approximation may be used to calculate both the radiation force and torque. The approximation consists of integrating the monopole and dipole contributions to the force throughout the volume of the object, and thus it is applicable to objects with arbitrary shapes and material property distributions. Here, an axisymmetric object in a plane standing wave is considered, resulting in one-dimensional integral expressions for the radiation force and torque. The integrals are evaluated analytically for homogeneous spheres and cylinders. The accuracy of the approximation is assessed via comparison with full solutions for spheres and prolate spheroids based on eigenfunction expansions for the corresponding scattering problem. Different densities and compressibilities of the object and the surrounding medium, as well as different sizes, shapes, and orientations of the object relative to the standing wave field, are considered. Limitations of the approximation and potential extensions to more complex wave fields are discussed. [T.S.J. supported by the ARL:UT McKinney Fellowship in Acoustics.]

11:45

2aPA13. Simulation of the second harmonic ultrasound field in heterogeneous soft tissue using a mixed domain method. Juanjuan Gu and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., EB III, campus box 7910, Raleigh, NC 27695, jgu4@ncsu.edu)

A mixed-domain method dubbed frequency-specific mixed domain method is introduced for the simulation of the second harmonic ultrasound field in weakly heterogeneous media. The governing equation for the second harmonics is derived based on the quasilinear theory. The speed of sound, nonlinear coefficient, and attenuation coefficient are all spatially varying functions in the equation. The fundamental frequency pressure field is first solved by the frequency-specific mixed domain method, and it is subsequently used as the source term for the second harmonics equation. This equation can be again solved by the frequency-specific mixed domain method to rapidly obtain the second harmonic pressure field. Five two-dimensional cases, including one with a realistic human tissue map, are studied to systematically verify the proposed method. Results from the previously developed transient mixed domain method are used as the benchmark solutions. Comparisons show that the two methods give similar results for all cases. More importantly, the frequency-specific mixed domain method has a crucial advantage over the transient mixed domain method in that it can be two orders of magnitude faster.

Session 2aPPa

Psychological and Physiological Acoustics and Education in Acoustics: Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical Physiological Collaborations

Kelly L. Whiteford, Cochair
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Anahita H. Mehta, Cochair
University of Minnesota, N640, Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455

Chair's Introduction—8:00

Invited Papers

8:05

2aPPa1. Effects of noise-induced hearing loss on speech-in-noise envelope coding: Inferences from single-unit and non-invasive measures in animals. Satyabrata Parida (Weldon School of Biomedical Eng., Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, spsatyabrat@gmail.com) and Michael G. Heinz (Speech, Lang., and Hearing Sci. & Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Speech-intelligibility models (SIM) can be used for systematic fitting of hearing-aids and cochlear-implants, potentially improving clinical outcomes in noisy environments. Existing SIMs are suitable for predicting performance of normal-hearing subjects, but not for hearing-impaired subjects due to our limited understanding of the effects of cochlear hearing impairment on speech-in-noise coding. In this work, we collected auditory-nerve (AN) single-unit responses and envelope following responses (EFR) in normal- and hearing-impaired chinchillas to speech, spectrally-matched stationary-noise, and noisy-speech. Our data show increased correlation between AN-fiber response envelopes of noisy-speech and noise-alone for hearing-impaired fibers in speech-relevant modulation-frequency bands, suggesting a greater degree of distraction from inherent envelope fluctuations following cochlear hearing loss. This novel finding is significant given the emphasis recent SIMs [e.g., Jørgensen and Dau, *JASA* (2011)] have placed on the importance of inherent noise-envelope fluctuations in addition to speech-coding fidelity in predicting noisy-speech perception. Preliminary data also show enhanced fundamental-periodicity coding at the expense of place-specific formant coding, and a degradation of burst envelopes of high-frequency fricatives for the hearing-impaired group. EFRs show evidence for degraded tonotopic coding, as observed in single-unit responses [e.g., Henry *et al*., *J. Neurosci*. (2016)]. [Work supported by Action on Hearing Loss (UK).]

8:25

2aPPa2. Passive music listening: A modulation of resting-state functional connectivity to better dissociate tinnitus. Somayeh Shahsavarani, Yihsin Tai (Speech and Hearing Sci., Inst. for Adv. Sci. and Technol., Univ. of Illinois at Urbana-Champaign, 2514 Beckman 405 N Mathews Ave., Urbana, IL 61801, bahar@illinois.edu), Sara Schmidt, Rafay Khan (Neurosci. Program, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Fatima Husain (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Previous fMRI studies have shown tinnitus-related changes in resting-state functional connectivity (rs-fc) and have the potential for serving as biomarkers of tinnitus. In this study, we further investigated the effect of tinnitus on three intrinsic neural networks—the auditory network, the dorsal attention network (DAN), and the default mode network (DMN) while the participants were at rest or under a passive music listening condition. Our results indicated that listening to music altered the auditory network in both groups, with increased connectivity between inferior frontal gyrus and auditory areas in controls. This alternation was observed only in the patients with mild tinnitus handicap and was absent in the patients with severe tinnitus handicap. The effect of music on the DAN hinged upon hearing sensitivity: decreased connectivity between the lateral occipital cortex and the DAN, and increased connectivity between the precuneus and the DAN was observed in controls and patients with normal hearing, compared to those with hearing loss. Furthermore, passive music listening modulated the coherency of the DMN based on tinnitus status and/or hearing sensitivity. Our findings highlight the efficacy of rs-fc in dissociating the relatively heterogeneous tinnitus population and its subgroups from controls, using rest and listening to music.

8:45

2aPPa3. Evaluating human neural envelope coding as the basis of speech intelligibility in noise. Vibha Viswanathan (Weldon School of Biomedical Eng., Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, viswanav@purdue.edu), Hari M. Bharadwaj (Speech, Lang., & Hearing Sci. and Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN), Barbara Shinn-Cunningham (Neurosci. Inst., Ctr. for the Neural Basis of Cognition, Biomedical Eng., Psych., and Elec. & Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA), and Michael G. Heinz (Speech, Lang., & Hearing Sci. and Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Models of speech intelligibility that accurately reflect human listening performance across a broad range of background-noise conditions are clinically important (e.g., for deriving hearing-aid prescriptions, and optimizing cochlear-implant signal processing). A leading hypothesis in the field is that internal representations of envelope information ultimately determine intelligibility. However, this hypothesis has not been tested neurophysiologically. Here, we address this gap by combining human electroencephalography (EEG) with simultaneous perceptual intelligibility measurements. First, we derive a neural envelope-coding metric (*ENVneural*) from EEG responses to speech in multiple levels of stationary noise, and identify a mapping between the neural metric and corresponding speech intelligibility. Then, using the same mapping, we use only EEG measurements to test whether *ENVneural* is predictive of speech intelligibility in novel background-noise conditions and in the presence of linear and non-linear distortions. Preliminary results suggest that neural envelope coding can predict speech intelligibility to varying degrees for different realistic listening conditions. These results inform modeling approaches based on neural coding of envelopes, and may lead to the future development of physiological assays for characterizing individual differences in speech-in-noise perceptual abilities.

9:05

2aPPa4. Effect of noise reduction on cortical speech processing in hearing aid users. Subong Kim, Inyong Choi, and Yu-Hsiang Wu (Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, subong-kim@uiowa.edu)

Noise reduction (NR) has been widely used in hearing aids (HAs) to increase ease and comfort of listening and to reduce listening effort. However, NR attenuates noise at the potential cost of distorting speech cues. This makes it challenging for audiologists to select the best configuration for NR during HA fitting process. The long-term goal of our research is to optimize HA fitting by characterizing the neural mechanisms underlying the effect of NR. The purpose of the present study is to examine the effect of NR on cortical dynamics during speech-in-noise tasks in HA users using electroencephalography. Our recent study with normal-hearing listeners has shown that speech recognition in low-level noise engaged greater early activity (~300 ms after word onset) in left supramarginal gyrus and weaker late activity (~700 ms) in left inferior frontal gyrus, than in high-level noise. Based on these findings, we hypothesized that, for a given patient, the optimal NR configuration would be the one that can recruit this "low-level noise" pattern of neural activity. Initial results from the electroencephalographic source space analysis will be presented, and underlying cortical mechanisms of speech processing in HA users will be discussed.

9:25

2aPPa5. Displacement of the stapes differs across species—Implications for studies of auditory function. John Peacock, Mohamed Alhussaini, Nate Greene, and Daniel J. Tollin (Univ. of Colorado, 12800 E 19th Ave., Aurora, CO 80045, john.2.peacock@ucdenver.edu)

Sound is transferred to the cochlea via the middle ear. The anatomy and physiology of the middle ear varies significantly across species, and these differences impact both the stimulation provided to the inner ear, and the suitability of different animal models for use in various types of research. Studies of auditory trauma from blast, for example, require generation of intracochlear pressures with sufficiently high intensities to cause damage. In some species, e.g., mice and rats, it may not be possible to generate sufficiently high pressures through air conducted sound alone, whereas in humans sufficiently high pressures can readily be generated through air conduction. We hypothesize that this is due to limits on the displacement of the stapes by the stapedial annular ligament, which thereby constrains the energy transferred to the cochlea through the middle ear. To test this hypothesis, we made measurements of the motion of the middle ear bones in response to tones of varying intensities and frequencies in several different species commonly used in laboratory research. Our results reveal peak stapes displacements from ~150 um in humans to 10-20 um in mice and rats. We will discuss the implications of these findings for basic studies of auditory function.

9:45

2aPPa6. Simultaneous measurement of electrocochleography and ear canal pressure in normal-hearing adults. Jessica Chen, Michael Simpson, and Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1217 BEH S, Salt Lake City, UT 84112, jessica.chen@utah.edu)

The auditory system is regulated by various adaptive mechanisms that modify sound as it passes upstream to the cortex. Studying these adaptive mechanisms, such as the middle ear muscle (MEM) reflex and the medial olivocochlear (MOC) reflex, reveals how the normal auditory system adjusts to challenging listening tasks and how abnormal reflexes may contribute to perceptual difficulties experienced by patients with sensorineural hearing loss. Studies in laboratory animals reveal that MOC and MEM reflexes result in neural antimasking at the output of the auditory nerve. Antimasking effects at the level of the middle ear and at the level of the cochlear hair cells should carry over to effects observed at the output of the auditory nerve. Antimasking due to MEM and MOC reflexes has been difficult to study in humans since measurement approaches are limited to non-invasive techniques with only a single output variable (e.g., ear canal pressure). In these studies, we assess MEM and MOC function by simultaneous measurements of ear canal pressure and electrocochleography, including the cochlear microphonic potential and compound action potential. Simultaneous measurements are expected to reveal the antimasking effects of MEM and MOC reflexes to middle ear, hair cell, and auditory nerve responses.

Session 2aPPb

Psychological and Physiological Acoustics: Topics in Physiological and Psychoacoustics (Poster Session)

Gregory M. Ellis, Chair
Communication Sciences and Disorders, Northwestern University, Department of Communication Sciences and Disorders, Evanston, IL 60201

All posters will be on display from 9:00 a.m. to 12:00 noon. To give contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

2aPPb1. Effects of age, modulation rate, and modulation depth on sentence recognition in speech-modulated noise. Daniel Fogerty, Rachel E. Miller (Commun. Sci. and Disord., Univ. of South Carolina, 1224 Sumter St., Columbia, SC 29208, fogerty@sc.edu), Jayne B. Ahlstrom, and Judy R. Dubno (Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Speech is often heard in amplitude modulated backgrounds when speech is glimpsed during momentary masker dips. Increased masker modulation depth provides more improvement in the signal-to-noise ratio of glimpsed speech, but also improves noise modulation detection for potentially greater modulation masking. Mechanisms of modulation masking and glimpsing may also depend on the overlap of the modulation spectra of the speech and noise. This study employed modulation filtering and noise amplitude compression to investigate the combined effects of noise modulation depth and modulation rate on speech recognition. Younger normal-hearing (YNH), older normal-hearing (ONH), and older hearing-impaired (OHI) adults listened to sentences in noise that were spectrally shaped to control for individual hearing thresholds. A second YNH group listened to sentences with the same spectral shaping as OHI listeners. Sentence recognition generally improved with greater noise modulation depth, especially at higher modulation rates. Results suggest that speech recognition for all groups is maximized when speech modulations <8 Hz are preserved, when noise modulation occurs at rates higher than this range, and with greater noise modulation depth. OHI listeners benefit similarly from these conditions, but their poorer overall performance may be due to reduced sensation levels. [Work supported by NIH/NIDCD.]

2aPPb2. Psychophysical and anatomical evidence for hidden hearing loss in laboratory mice. Kali Burke (Psych., Univ. at Buffalo, SUNY, 246 Park Hall, Buffalo, NY 14260, kaliburk@buffalo.edu), Laurel A. Screven (Otolaryngol. – Head and Neck Surgery, Johns Hopkins Univ. School of Medicine, Baltimore, MD), Anastasiya Kobrina (Psych., Univ. at Buffalo, SUNY, Amherst, NY), Katrina M. Schrode, Amanda Lauer (Otolaryngol. – Head and Neck Surgery, Johns Hopkins Univ. School of Medicine, Baltimore, MD), and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, Buffalo, NY)

Exposure to high intensity sound can lead to temporary or permanent threshold shifts. Noise exposures that do not cause long-term hearing deficits, however, can induce extensive afferent ribbon synapse loss, while hair cells and spiral ganglion neurons remain mostly intact (Kujawa and Liberman, 2009). This loss of synapses despite normal hearing thresholds is referred to as hidden hearing loss (HHL). We examined the development of HHL in laboratory mice using operant conditioning with positive reinforcement. After exposing mice to 8–16 kHz narrowband noise at 100 dB SPL for 2 h, hearing thresholds temporarily shifted for both pure tone and ultrasonic vocalization stimuli; however, post-exposure thresholds and threshold shifts varied by sex and age. Immunohistochemistry and transmission electron microscopy were conducted to quantify peripheral damage and central synaptic reorganization once behavioral testing was complete. Brains were collected, sectioned, and labeled against VGLUT1 or GAD65 and labeling was quantified in the ventral cochlear nucleus. Cochleas were also collected, dissected, and labeled for myosin6 to label hair cells and either CTBP2 or SV2 to identify afferent or efferent terminals, respectively. Our findings show that mice are able to behaviorally recover hearing following non-traumatic noise exposure despite changes in peripheral and central auditory structures.

2aPPb3. The effect of broadband elicitor duration on transient-evoked otoacoustic emissions and a psychoacoustic measure of gain reduction. William Salloom (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, wsalloom@purdue.edu), Hari M. Bharadwaj, and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci. and Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Physiological and psychoacoustic studies of the medial olivocochlear reflex (MOCR) in humans have often relied on long elicitors ($>$ 100 ms). This is largely due to previous research using otoacoustic emissions (OAEs) that found MOCR time constants in the 100s of milliseconds when elicited by broadband noise. However, Roverud and Strickland (2014), using a psychoacoustic measure of gain reduction, found differential effects of duration for on- and off-frequency tonal elicitors. For the on-frequency elicitor, thresholds increased with increasing on-frequency duration up to about 50 ms, and then plateaued. In contrast, thresholds with off-frequency elicitors continued to increase with elicitor duration. These results are consistent with cochlear gain reduction, possibly by the MOCR, in which the on-frequency elicitor is affected by gain reduction at the signal frequency place, but the off-frequency elicitor is not. The effect of the duration of broadband noise elicitors on similar psychoacoustic tasks is currently unknown. Additionally, the relationship between gain reduction measured psychoacoustically and using OAEs as a function of elicitor duration are unknown. This study will measure the effects of ipsilateral broadband noise elicitor duration on transient-evoked OAEs and psychoacoustic gain reduction estimated from a forward-masking paradigm. [Work supported by NIH R01 DC008327 (EAS) and NIH R01DC015989 (HMB).]

2aPPb4. Physiological studies of binaural tone-in-noise detection in the inferior colliculus of awake rabbit. Langchen Fan, Kenneth S. Henry, and Laurel H. Carney (Univ. of Rochester, MC 5-7616 601 Elmwood Ave., Rochester, NY 14642, langchen.fan@gmail.com)

With identical noise maskers presented to both ears, human listeners have lower tone detection thresholds when the tone is presented out-of-phase between the ears (N_0S_π) rather than in-phase (N_0S_0). The threshold difference is called the binaural masking level difference (BMLD). Human listeners have BMLDs up to at least 4 kHz, but previous studies to understand neural mechanisms have only used low-frequency tones. Here, we recorded single-neuron responses from the inferior colliculus in awake rabbit to a wide range of overall noise levels and tone frequencies near the neuron's characteristic frequency. Maskers were 1/3-octave gaussian noise centered at the tone frequency. Neural thresholds at each noise level were estimated from average-rate responses at signal-to-noise ratios of -12 to +8 dB based on receiver-operating-characteristic analysis. Neural thresholds were similar across noise levels and were pooled together for further analysis. BMLDs of all recorded neurons with measurable thresholds for both N_0S_0 and N_0S_π ranged up to +17 dB (i.e., a substantial masking release for the N_0S_π condition). Across the population of neurons tuned to different frequencies, the largest BMLDs at each frequency decreased with increasing frequency, consistent with human psychophysical studies, but with a more gradual slope.

2aPPb5. Influence of viscous boundary layer on wave propagation on isolated tectorial membranes. Haiqi Wen, Charlsie Lemons, Elisa Boatti, and Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta 30339, GA, hwen@gatech.edu)

The tectorial membrane is an extracellular matrix located in the cochlea. The tectorial membrane is often hypothesized to play an important role in hearing mechanics. Measurements of wave propagation on isolated tectorial membranes have been used in the literature to characterize the intrinsic mechanical properties of tectorial membranes in the auditory frequency range. While most previous studies have made an implicit assumption regarding the width of the tectorial membranes in order to find the properties of the TM using a simple model, we have recently used a more accurate model that takes into account the finite width of the TM and its anisotropy. However, experiments are conducted in an artificial endolymph bath, which we neglected in our previous analysis. In this work, we study the influence of the viscous boundary layer due to the fluid on wave propagation on isolated tectorial membranes. The boundary layer adds damping and mass to the tectorial membrane, which we model using a commercial finite element software. The influence of the boundary layer on the space constant and wave speed of the longitudinally propagating radial motion, and on the spatial patterns of the longitudinal motion, are analyzed.

2aPPb6. Implementing an efferent signal into an auditory pathway model for tone-in-noise detection. Afagh Farhadi (Elec. and Comput. Eng., Univ. of Rochester, 1900 Wind Willow Way Apt. 11, Rochester, NY 14624, afarhadi@ur.rochester.edu) and Laurel H. Carney (Biomedical Eng. & Neurosci., Univ. of Rochester, Rochester, NY)

Simultaneous tone-in-noise detection has been studied extensively, but typically without consideration of the medial olivocochlear (MOC) efferents. We are testing hypotheses for masked detection using a central auditory model with a signal from midbrain to MOC. Masked tones are encoded in the rate profile of band-enhanced (BE) inferior colliculus (IC) neurons, which are excited by a range of modulation frequencies. Peripheral responses to noise are characterized by large fluctuations, an effective stimulus for BE IC neurons. In contrast, peripheral channels tuned near the tone have smaller fluctuations: addition of the tone flattens the signal envelope and also pushes the inner hair cell (IHC) transduction nonlinearity further into saturation. Excitatory projections to MOC from noise-driven BE IC cells would decrease cochlear gain, reducing IHC saturation, and resulting in larger fluctuations and IC rates. In contrast, tone-plus-noise-driven channels would reduce MOC excitation, resulting in relatively higher cochlear gain, more saturation, and ultimately lower IC rates. Thus, the descending signal from IC BE cells to MOC is hypothesized to enhance contrast in the IC rate profile. Because efferents have slow dynamics, timing is an important factor. Therefore, we focus on model sensitivity to masked tones of different durations for comparison to psychophysical trends.

2aPPb7. Detection and intensity discrimination of a sinusoid with and without masker uncertainty. Christopher Conroy, Christine Mason, and Gerald Kidd (Dept. of Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cwconroy@bu.edu)

The effect of stimulus-based uncertainty on the shape of the masking function at circa-threshold pedestal levels was investigated. Of particular interest was how uncertainty regarding properties of the masker affects the magnitude of "negative masking" [Raab *et al*., *J. Acoust. Soc. Am.* **35**, 1053 (1963)] that is obtained under different stimulus configurations. Intensity discrimination thresholds for gated, 100-ms, 1000-Hz sinusoids were measured at three pedestal levels: -9, 0, and 9 dB re: absolute threshold. A two-interval, two-alternative forced-choice procedure was used. Under the reference condition, thresholds were measured in quiet. In comparison conditions, thresholds were measured in the presence of one of two masker types: (1) a notched-noise masker or (2) a random-frequency, multicomponent masker. In the multicomponent masker condition, uncertainty was imposed by roving the frequency components that comprised the masker from interval-to-interval over a block of trials. The data under each condition were fit with psychometric functions and slope estimates were obtained. Results are discussed with respect to the factors that govern, and possible mechanisms underlying, negative masking. Specifically, two possible explanations for the "dip" in the masking function are examined: nonlinear transduction [Hanna *et al*., *J. Acoust. Soc. Am.* **80**, 1335 (1986)] and observer uncertainty.

2aPPb8. Sensitivity to AM incoherence is affected by center frequency and modulation rate. Kelly L. Whiteford and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu)

Fine-grained sensitivity to frequency modulation (FM) at slow rates and low-frequency carriers is thought to be due to auditory-nerve phase locking (time code). Alternatively, a unitary code for FM at all rates and carrier frequencies could be based on cochlear conversion of FM to amplitude modulation (AM) (place code). One weakness of the place-coding theory is it cannot readily explain rate- and carrier-dependent trends in FM sensitivity. This study asked whether FM trends could potentially be explained by sensitivity to two AM envelopes that are out of phase (incoherent AM) at separate cochlear locations, thereby simulating the effects of FM. AM discrimination was assessed for two-component complexes centered at low (500 and 1500 Hz) and high (7000 Hz) frequencies, spaced 2/3 or 4/3 octaves apart, and modulated at slow (2 Hz) and fast (20 Hz) rates. Coherent and incoherent two-component AM detection was assessed for the same conditions. Preliminary results show that sensitivity to AM incoherence is best at low center frequencies and slow rates, consistent with trends traditionally found in FM detection that have been attributed to time coding. Findings suggest time coding may not be necessary to explain trends in FM sensitivity. [Work supported by NIH Grant R01DC005216.]

2aPPb9. Discrimination of rippled spectra: Contribution of excitation-pattern and temporal-processing mechanisms. Olga Milekhina, Dmitry Nechaev, and Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru)

Rippled-spectrum signals are used for measurements of signal resolution in hearing-impaired listeners and cochlear-implant users. Two mechanisms may be responsible for rippled-spectrum resolution. The excitation-pattern mechanism determines the ripple density resolution (ripples/oct). The temporal-processing mechanism determines the ripple frequency spacing limit (kHz). Contributions of the mechanisms can be assessed by comparison of resolutions of band-limited rippled spectra with different center frequencies, because the ratio of rippe spacing to ripple density is frequency-proportional. Ripple-density resolutions of half-octave rippled spectra were measured at center frequencies from 0.5 to 4 kHz. The

measurements were performed either by discrimination between rippled-spectrum test and reference signals differing by ripple phases or by discrimination between a rippled-spectrum test and non-rippled reference signal. For discrimination between rippled-spectrum test and reference signals, resolution specified in ripples/oct little depended on center frequency, as predicted by the excitation-pattern model. For discrimination between rippled test and non-rippled reference signals, the resolution specified in ripple frequency spacing little depended on center frequency, as predicted by the temporal-processing model. It was concluded that contributions of the excitation-pattern and temporal-processing mechanism depend on the discrimination task.

2aPPb10. Rapid measurement of frequency depending spectro-temporal modulation detection thresholds. Stefan Klockgether, Julia Rehmann, and R. Peter Derleth (R&D, Sonova AG, Laubisrütistrasse 28, Stäfa, Zürich 8712, Switzerland, stefan.klockgether@sonova.com)

The modulation detection threshold of spectro-temporal modulations (STM) is highly correlated with the speech reception threshold (SRT). A common interpretation of this correlation is that detection capability of concurrent spectral and temporal modulations is a measure for the frequency resolution power of the auditory system. The STM-detection threshold can be measured for different frequency regions by applying the modulation only to particular frequency bands. One could make an estimation of the frequency resolution power at those particular frequency regions. This could give a hint on a beginning or a hidden hearing loss which is not showing up clearly in an audiogram. This study shows different methods to rapidly measure STM-detection thresholds frequency dependent and compares the results with results gained with an established adaptive staircase method. The rapid methods are evaluated with regard to their practicability and the trade-off between reduced duration and reduced accuracy.

2aPPb11. The duplex theory re-revisited: Spectral weighting of localization cues in tones and noises. G. Christopher Stecker, Monica L. Folkerts, and Julie M. Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu)

It is widely suspected that sound localization is accomplished primarily through the use of low-frequency interaural-time-difference (ITD) cues and only secondarily via high-frequency ITD and level-difference (ILD) cues [Rayleigh, *Philos. Mag.* **13**, 214–232 (1907); Wightman and Kistler, *JASA* **91**, 1648-1661 (1992); Macpherson and Middlebrooks *JASA* **111**, 2219–2236 (2002)]. Contemporary studies of cross-frequency interactions in spatial hearing have provided support for this view but have not directly identified the frequencies involved nor quantified the relative weighting of binaural cues weighting across components of a single complex. This study adapted the temporal-weighting approach of Stecker and Hafter [*JASA* **112**, 1046–1057 (2002)] to measure spectral weighting functions (SWF) for (a) free-field and reverberant sound localization and (b) lateralization based on ITD and/or ILD. Across a wide range of conditions, SWFs featured enhanced weights for components in the vicinity of 400-800 Hz, supporting a narrow "dominance region" for localization and lateralization of complex sounds [Bilsen and Raatgever, *Acoustica* **28**, 131–132 (1973)], and a precipitous drop in weights between 800 and 1400 Hz [Brughera *et al.*, *JASA* **133**, 2839–2855 (2013)]. [Work supported by NIH R01-DC016643.]

2aPPb12. Consequences of interaural decorrelation of speech temporal fine structure. Lucas Baltzell, Virginia Best, and Jayaganesh Swaminathan (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, lbaltzel@bu.edu)

The ability to understand a target speech signal against a background of interfering speech signals is typically improved when the interfering signals are spatially separated (spatial release from masking; SRM). Swaminathan *et al.* (2016) found a significant reduction in SRM when the temporal fine structure (TFS) across the left and right ears was decorrelated, suggesting that binaural TFS provides important cues that support SRM. One interpretation is that degrading binaural TFS prevents the extraction of reliable interaural time differences (ITDs), which in turn leads to a reduction in SRM. We tested this hypothesis by systematically decorrelating the binaural TFS, and measuring the effects on both ITD discrimination for speech stimuli and SRM in normal hearing listeners. We show that decorrelation leads to a systematic increase in ITD discrimination thresholds, as well as a systematic decrease in SRM. This supports the idea that binaural TFS is needed to access ITD information in speech, which is in turn required for SRM. Additionally, the relationship between these two tasks provides a framework for determining the extent to which reduced SRM in listeners with hearing loss can be attributed to reduced sensitivity to binaural TFS.

2aPPb13. Decoding listener's attention :Can it be improved with behavioural measure of selective attention? Moïra-Phoebé Huet (Univ. of Lyon, 25 Ave. Jean Capelle O, LVA, Villeurbanne 69100, France, moira-phoebe.huet@insa-lyon.fr), Christophe Micheyl (Starkey, Lyon, France), Etienne Gaudrain (Univ. of Lyon, Groningen, The Netherlands), and Etienne Parizet (Univ. of Lyon, Villeurbanne, France)

During the past decade, there has been growing interest in the neural correlates of selective attention to speech. In these studies, listeners were instructed to focus their attention on one of two concurrent speech streams. However, in everyday-life situations, a listener's attention can switch rapidly between different voices. Thus, we have developed a behavioural protocol to infer the dynamics of auditory attention over time. After listening to two simultaneous stories—a target and an interferer—the participants have to find, among a set of words, those present in the target story. The participant's responses are then used to estimate, retrospectively, when their attention was directed toward the target, or toward the interferer. Neural data, recorded with EEG, and behavioural measures are combined to extract the brain's temporal response function in response to these stimuli. Moreover, to promote attention switches between the two voices, the interferers were uttered by the same talker as the target stories, but the voice parameters were manipulated to parametrically control the similarity of the two voices. We will discuss the results in terms of attentional selection and voice confusion, and suggest possible applications of this dynamic behavioral test of selective auditory attention.

2aPPb14. Ability of normal hearing listeners to recognize vowels and musical instruments under spectrally-degraded conditions. Ryan Anderson, Alyxandria Sundheimer, and William Shofner (Speech and Hearing Sci., Indiana Univ., 200 S Jordan, Bloomington, IN 47401, anderyan@indiana.edu)

Music perception requires greater spectral resolution than speech perception [Shannon, *Int. Rev. Neurobiol.* **70**, 121 (2005)]. However, conclusions from these metadata are problematic given that they aggregate results from several different studies using diverse methodologies and paradigms. In particular, methodologies generally tapped into different perceptual dimensions, namely word recognition for speech and melody recognition for music. The present study aims to develop a paradigm to compare speech and music recognition based on similar perceptual dimensions, namely timbre. Stimuli consisted of naturally-spoken vowels and notes played on musical instruments as well as 32-, 8-, and 4- channel noise-vocoded versions. Listeners discriminated either instruments from vowels or vowels from instruments using a go/no-go task. The discrimination paradigm offered insight as to how available spectral information influenced perception between stimulus sound categories. Reaction times and accuracy were measured and organized as a function of signal degradation level to consider potential differences in how normal hearing listeners utilize spectral fine structure information when distinguishing vowels and musical instruments. In general, listeners' ability to discriminate vowels from instruments under degraded conditions was similar to their ability to discriminate instruments from vowels. The results suggest that the perception of vowels and musical instruments rely on similar mechanisms.

2aPPb15. Pure-tone thresholds and the binaural masking level difference. Daniel E. Shub, Joshua G. W. Bernstein, Lina R. R. Kubli, Douglas S. Brungart (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 4954 N. Palmer Rd., Bethesda, MD 20889, joshua.g.bernstein.civ@mail.mil), and Ken W. Grant (National Military Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Silver Spring, MD)

Although several studies have examined the relationship between high-frequency pure-tone thresholds and the 500-Hz binaural masking level difference (BMLD), the results have not always been consistent. In this study, a retrospective analysis was conducted on an existing dataset from over 3000 military service members that included both pure-tone thresholds measured as part of their annual hearing conservation testing and a 33-trial clinical test of the BMLD. There was only a slight dependence of the 500-Hz BMLD on high-frequency pure-tone thresholds. For listeners with elevated pure-tone thresholds, this dependence was in good agreement with the findings of Jerger *et al*. [*Arch. Otolaryngol*. **110**, 290–296 (1984)] and slightly larger than that reported by Wilson and Weakley [*J. Am. Acad. Audiol*. **16**, 367–382 (2005)]. For listeners with near-normal hearing, the dependence of the 500-Hz BMLD on the 4-kHz pure-tone threshold was substantially less than that reported by Bernstein and Trahiotis [*J. Acoust. Soc. Am*. **140**, 3540–3548 (2016)]. A possible explanation might be the degree of training offered to the subjects and procedural differences between clinical and laboratory techniques. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

2aPPb16. The fine-grained statistical structure of speech is congruent with nonlinear peripheral auditory processing. François Deloche (CAMS, PSL Univ., 54 Blvd. Raspail, Paris 75006, France, francois.deloche@ehess.fr)

Efficient coding of sensory signals takes advantage of statistical regularities in sensory data. Cochlear filter in mammals are known to reflect the overall statistical structure of speech, in line with the hypothesis that low-level sensory processing provides efficient codes for information contained in natural stimuli. Recently, some efforts have been made to describe this correspondence in more detail. The study of the statistical structure of speech over different acoustic classes demonstrates that frequency selectivity should not be fixed to achieve maximum efficiency. On the other hand, cochlear signal processing is nonlinear as frequency selectivity decreases with sound intensity level. Both effects are greater in the high frequencies. In the present study, these two facts are shown to be consistent in the case of a parametric method based on Gabor dictionaries (Gaussian-modulated sinusoids) and in a simplified setting. A model with fewer constrains is also introduced for future experiments to validate this hypothesis in a more general context.

2aPPb17. Harmonicity aids detection of speech and other sounds in noise. Malinda J. McPherson (Div. of Medical Sci., Harvard Univ., MIT Bldg. 46-4078, 43 Vassar St., 46-4078, Cambridge, MA 02139, malindamcpherson@g.harvard.edu) and Josh McDermott (Dept. of Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA)

Acoustic grouping cues, such as the tendency of frequencies to be harmonic, are used to segregate multiple sounds, as when listening to one of several concurrent speakers. However, we often must listen to sound sources in noise. Here we investigate the role of traditional acoustic grouping cues in detecting sound sources in noise. We measured detection thresholds for several types of sounds embedded in noise: speech, musical instruments, and synthetic complex tones. In each case the sounds were resynthesized with harmonic and inharmonic carrier frequencies to test the importance of harmonic frequency structure for hearing in noise. We found that harmonic signals were consistently easier to detect in noise than otherwise similar inharmonic signals. This harmonic advantage persisted even when the phases of harmonic components were randomized, such that the advantage is unlikely to reflect differences in the depth of modulation (from beating). Near threshold, harmonic signals were readily audible in conditions where inharmonic signals could not be heard. These results suggest that harmonicity is critical for detecting real-world signals in noise, demonstrating its relevance to another important aspect of auditory scene analysis.

2aPPb18. Does lateral position explain release from informational masking arising from interaural time and level differences? Richard L. Freyman, Casey D. Milkey (Dept. of Commun. Disord., Univ. of Massachusetts, Amherst, MA 01003, rlf@comdis.umass.edu), Emily Buss (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Patrick Zurek (Sensimetrics Corp., Malden, MA)

Spatial release from masking (SRM) is known to arise from the interaural time and level differences (ITDs and ILDs) produced by spatially separated signal and masker sources. In some situations, SRM might be augmented considerably by the differing spatial perceptions created by the signal and masker, which could help alleviate confusions between them. Using headphone presentation, the present study investigated this hypothesized perceptual component of SRM. Maskers were binaural click trains of approximately 2-s overall duration with inter-click intervals (ICIs) varying randomly between 50 and 150 ms. The ITD or ILD of the masker clicks was manipulated across listening blocks. The listeners' task was to identify the number of signal clicks, identical to the masker clicks except for their ITD/ILD, inserted during masker inter-click intervals. Preliminary testing with a fixed masker ICI demonstrated that the signal clicks were easily audible, strongly suggesting that difficulty during the main identification task was due to confusions from the randomized ICI. The study explored whether performance counting signal clicks in the randomized ICI conditions could be explained by the differences in intracranial spatial perception created by combinations of ITD and ILD cues. [Work supported by NIDCD R01 01625.]

2aPPb19. Revisiting superoptimal perceptual integration for pitch at high frequencies. Hedwig E. Gockel and Robert P. Carlyon (MRC Cognition and Brain Sci. Unit, Univ. of Cambridge, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, hedwig.gockel@mrc-cbu.cam.ac.uk)

In a frequency discrimination task, Lau *et al*. [*J. Neurosci*. (2017)] reported superoptimal integration of information from individual components in a very high frequency region, where phase locking to the temporal fine structure is presumably absent, when the components were combined into a harmonic complex. We tried to replicate this finding using stimuli identical to those of Lau *et al*., with some additional conditions. Using an adaptive two-alternative forced-choice procedure, we measured fundamental frequency difference limens (F0DLs) for complex tones containing harmonics 6–10 with F0s of 280 and 1400 Hz, and frequency difference limens (FDLs) for each harmonic of the complex presented alone. Stimulus duration was 210 and 1000 ms. All tones had random phases, a ± 3 dB level rove, and were presented in a continuous threshold-equalizing noise that was either diotic or dichotic. Observed DLs were lower overall than in the study of Lau *et al*. As in their study, for the low F0, observed F0DLs were worse than predicted assuming optimal combination of frequency information from the individual harmonics and assuming that performance is limited by peripheral noise. However, for the high F0, observed and predicted F0DLs did not differ significantly, in contrast to the finding of Lau *et al*.

2aPPb20. Age-related differences in modulation detection interference and interference release. Yuan He and Jennifer Lentz (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, heyuan@indiana.edu)

Age-related declines have been observed in auditory tasks related to temporal processing, but less work has been conducted in middle-aged listeners and in auditory sound segregation tasks. This experiment addresses

these issues by evaluating modulation detection interference (MDI) and release from MDI using streaming in young, middle-aged and older listeners. Using a standard MDI paradigm, we measured amplitude modulation detection thresholds of a high-frequency tone in different conditions: in quiet, in the presence of a low-impact unmodulated interferer, and in the presence of a high-impact modulated interferer. Then, we measured streaming-based release from MDI using four modulated precursors designed to perceptually capture the interferer and therefore reduce the amount of interference. Preliminary data suggest that middle-aged normal-hearing listeners experience similar MDI to young listeners and also receive a large release from interference when precursors are present. Additional data will be presented from older listeners, who experience slightly greater MDI than younger listeners.

2aPPb21. Blast exposure in the military and its effects on sensory and cognitive auditory processing. Scott Bressler (Biomedical Eng., Boston Univ., 610 Commonwealth Ave., Rm. 923C, Boston, MA 02215, bressler@bu.edu), Kimberly Jenkins, Jennifer Myers, Kenneth Grant (Audiol., Walter Reed National Military Medical Ctr., Bethesda, MD), and Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA)

Blast-induced traumatic brain injury (TBI) and hearing loss are the two most common types of injuries sustained by military personnel while serving in the U.S. Global War on Terrorism. Recently several VA audiology clinics have reported active duty service members complaining of having problems communicating in noisy listening environments despite having normal to near-normal pure tone thresholds. In addition to standard clinical measures, we used electroencephalography (EEG) to determine whether damage to suprathreshold responding auditory nerve fibers in the sensory periphery and/or trauma to cortical regions associated with attention and working memory were responsible for the reported listening complications. In separate auditory and visual selective attention tasks, behavioral and neural measures suggest no evidence of long term neurotrauma affecting normal cognitive function. We found while absolute measures of auditory brainstem encoding varied greatly in all study subjects, comparisons of how the envelope following response (EFR) changes with modulation depth hint at differences between blast and non-blast exposed service members. These findings are consistent with audiometric threshold and distortion product otoacoustic emission data that show subtle differences between groups within clinically defined normal limits. Taken together these results suggest subclinical differences in audiometric measures might explain differences in suprathreshold listening.

2aPPb22. A harmonic-cancellation-based model to predict speech intelligibility against a harmonic masker. Luna Prud'homme, Mathieu Lavandier (Laboratoire Génie Civil et Bâtiment, Univ Lyon, ENTPE, rue Maurice Audin, Vaulx-en-Velin 69518, France, luna.prudhomme@entpe.fr), and Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

This study aimed at predicting speech intelligibility in the presence of harmonic maskers. Contrary to a noise signal, these maskers have a harmonic structure that allows for a segregation of the competing sounds based on a difference of their fundamental frequency (F0). This F0 segregation could be due to spectral glimpsing or harmonic cancellation. It is unclear what the relative contributions of these two mechanisms are. In this work, we have modified the model of Collin and Lavandier [*J. Acoust. Soc. Am.* **134**, 1146–1159 (2013)] in order to take into account both mechanisms. Different implementations of the model were compared and applied to two data sets: SRTs measured for monotonous harmonic maskers that varied in their F0s and degree of harmonicity; SRTs measured for monotonous and intonated harmonic maskers that varied in their mean F0, F0 contour, temporal envelope and spatial position. Comparison of data and model predictions will allow us to establish how to implement harmonic cancellation in our model framework, to evaluate to what extent the model can predict F0 segregation for harmonic maskers, and to determine the relative roles of spectral glimpsing and harmonic cancellation suggested by the model.

2aPPb23. Backward recognition masking of similar and dissimilar environmental sounds. Kelli Clark, Allison Holtz, Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, kclark44@students.towson.edu), Jeremy Gaston, and Brandon Perelman (Human Res. and Eng. Directorate, U.S. Army Res. Lab., Adelphi, MD)

Backward Recognition Masking (BRM) of sound occurs when two sounds are presented close in time to one another and the second sound hinders the recognition of the first sound. Previous studies on BRM used either white noise or sine tones. Here, we present backward recognition thresholds using environmental sounds. 34 young normal hearing individuals were presented with 6 combinations of similar or dissimilar environmental sounds from the categories of household appliances, automobiles, and power tools. The sounds were selected based on their multi-dimensional scaling distances (Rosen *et al.*, 2017). The sounds were presented at 20 dB SL re: 3 frequency PTA and the inter stimulus interval was fixed at 22.2 ms. A two-down one-up adaptive procedure was used to identify the target duration at which the listeners could identify that the target is different from the masker. Initial data analyses indicated that similar sounds needed significantly longer presentation durations to be identified as different compared to dissimilar sounds, indicating more backward masking for similar sounds compared to dissimilar sounds. These results potentially highlight how critical safety information in real-world environments may be missed in complex listening scenarios.

2aPPb24. The effects of modulator shape and methods for expressing modulation depth on spectral modulation detection thresholds. Sittiprapa Isarangura, Katherine Palandrani (Univ. of South Florida, 4202 E. Fowler Ave. PCD1017, Tampa, FL 33620, sisarangura@mail.usf.edu), Trevor Stavropoulos, Aaron Seitz (Univ. of California, Riverside, Riverside, CA), Eric C. Hoover (Dept. of Commun. Sci. and Disord., Univ. of Maryland, College Park, MD), Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., Portland, OR), and David A. Eddins (Univ. of South Florida, Tampa, FL)

The detection of sinusoidal modulation is commonly used for assessing the auditory perception of temporal, spectral, and spectro-temporal acoustic features. For temporal (amplitude) modulation, the sinusoidal modulator usually is expressed on a linear amplitude scale. For spectral modulation, the sinusoidal modulator has been specified on a linear amplitude scale, consistent with temporal modulation, or on a logarithmic amplitude scale, with the notion of approximating a sinusoidal excitation pattern. The definition of modulation depth depends on the measurement points (i.e., midpoint to peak or peak to peak) and order of operations when expressing depth in dB. Such differences can make it difficult to compare results for similar tasks among investigations. Here we quantify differences among methods and provide a complete matrix for translating among methods. Spectral modulation detection was measured for 9 normal-hearing listeners in ten conditions (linear vs. logarithmic shape at 0.5, 1, 2, 4, and 8 cycles/octave). Peak-to-peak values of the modulation envelope were equalized, thus only modulation waveform shape differed. Noise carriers had a passband from 400 to 3200 Hz. Thresholds revealed statistically significant effects of both spectral shape and spectral modulation frequency. Several methods for expressing threshold modulation depth were compared to highlight differences among the methods.

2aPPb25. Pre-trial and post-trial cueing of masker location in a localization-in-noise task. Brian Simpson (Air Force Res. Lab., 2610 Seventh St., Area B, Bldg. 441, Wright-Patterson AFB, OH 45433, brian.simpson.4@us.af.mil), Robert H. Gilkey, Michelle Wang (Dept. of Psych., Wright State Univ., Dayton, OH), Nathaniel Spencer, and Eric R. Thompson (Air Force Res. Lab., Wright-Patterson AFB, OH)

Localization accuracy for a target presented in a simultaneous masker, whose location varies randomly from trial to trial, improves when a preview

of the masker location is provided (by playing a sound from that location) prior to the target + masker interval (i.e., a pre-trial cue) [B. Simpson, Ph.D. dissertation (Wright State University, 2011)]. One explanation is that knowing the masker location allows a listener to establish a "spatial attention filter" at the masker location. The present study compares the effect of such a pre-trial cue to the case in which the cue comes *after* the target + masker interval (post-trial cueing). That is, the cue is presented either 500 ms prior to the onset of a 60-ms, 100-Hz click-train target embedded in a 60-ms broadband masker, or 500 ms subsequent to the offset of the target + masker stimulus. The data indicate that both cue types lead to similar improvements in performance over the no-cue condition, with the greatest improvement from cueing (~6 dB) seen for localization in the left/right dimension. While these data are roughly consistent with previous results, they cannot be explained by a simple spatial filter hypothesis.

2aPPb26. The effect of simulated Doppler frequency changes on detectability of complex tones. Lawrence L. Feth, Evelyn M. Hoglund, Omkar Dixit (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, feth.1@osu.edu), and Matthew Davis (CDO Technologies, Columbus, OH)

The detectability of frequency changes designed to simulate the Doppler effect for moving sound sources has been well documented in the literature (Chowning, 1977; Ericson, 2001; Kaczmarek, 2005, Getzmann, 2008, and Pörschmann & Störig, 2009). The current study was designed to determine whether the presence of Doppler frequency shifts have a measurable effect on the detectability of complex sounds. Detection thresholds for single sinusoids with frequency changes ranging from the frequency DL to a full octave are compared with those for complex tones with selected components frequency modulated to simulate Doppler effects. A one-interval, adaptive tracking paradigm using the SIAM procedure (Kaernbach, 1991) was used to determine three points on the psychometric functions so that detection thresholds and slopes could be compared. For single component signals, the direction of frequency change and the span of the frequency sweep appear to have an effect on detectability. Results for complex tones require further explanation. [Work supported by a contract from CDO Technologies, Dayton, OH.]

2aPPb27. Masking of short tones in noise: Evidence for envelope-based, rather than energy-based detection. Skyler G. Jennings and Jessica Chen (Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Salt Lake City, UT 84112, skyler.jennings@hsc.utah.edu)

A short tone added to a longer simultaneous masker may be detected by observing an increase in overall acoustic energy. This detection scheme predicts probe thresholds to be independent of the masker's temporal envelope. A recent study [Jennings *et al*., *J. Assoc. Res. Oto.* **19**, 717–727 (2018)] revealed that probe thresholds were 10 dB higher for maskers with flattened compared to fluctuating envelopes, suggesting an envelope-based detection strategy. This study test the hypothesis that probe thresholds are proportional to envelope power by measuring detection thresholds for a 4-kHz, 6-ms probe in one of three temporal positions within a 400-ms masker in normal-hearing listeners. The narrow-band, low-fluctuating noise masker was preceded by flattened or fluctuating noise precursors. The precursor's offset was delayed from the masker's onset by -2, 0, 10, 25, 100, or 250 ms. Probe thresholds were positively correlated with envelope power (R-squared = 0.75), consistent with masking from precursor envelope fluctuations and from fluctuations introduced by precursor and masker ramps. These findings suggest that future modeling efforts for short probes presented in longer maskers should consider a decision variable based on envelope fluctuations.

2aPPb28. Surveying the sounds used in the Journal of the Acoustical Society of America (1950–2017). Michael Schutz and Jessica Gillard (School of the Arts, McMaster Univ., 1280 Main St. West, Hamilton, ON L8S4M2, Canada, schutz@mcmaster.ca)

The earliest auditory psychophysical experiments involved naturalistic sounds such as hammers striking plates. The subsequent development and ubiquity of desktop computing gave researchers the ability to more precisely control stimulus parameters such as frequency, amplitude, and duration (Neuhoff, 2004). However much of our everyday listening is for events rather than easily manipulated properties (Gaver, 1993), and the world lacks the kinds of constrained sounds often used in auditory research (Phillips *et al*., 2002). Although simplistic auditory stimuli hold benefits with respect to control, their disproportionate use poses problems for generalizing outcomes from key experiments. To provide insight into the sounds used in auditory perception research, we surveyed a representative sample of auditory stimuli from 217 psychophysical experiments published in JASA between 1950 and 2017. Our survey documents a disproportionate focus on simplistic sounds, with less than 4% of psychophysical experiments using stimuli exhibiting the dynamic temporal structures characteristic of natural auditory events. We will discuss the implications of these findings in the content of ongoing areas of inquiry of broad relevance to the auditory perception community.

2aPPb29. Auditory detection and sound source elevation. M. Torben Pastore and William Yost (College of Health Solutions, Arizona State Univ., 954 East Lobster Trap Ln., Tempe, AZ 85283, m.torben.pastore@gmail.com)

The spectral profile of a given sound stimulus varies with the spatial location according to the head-related transfer functions (HRTF) at both ears. For narrowband and tonal stimuli at frequencies above approximately 4 kHz, perceived sound source elevation has been shown to depend mainly on stimulus frequency instead of sound source location (e.g., Blauert, Morimoto), and this result has in turn been shown to correlate with spectral features of the HRTF (e.g., Middlebrooks). However, even though the HRTF is important to perceived elevation, listeners do not generally report a perceptual awareness of the spectral filtering that occurs as a result of the HRTF, suggesting that some form of spectral equalization may occur at some level beyond peripheral auditory processing. Therefore, depending on the level at which detection processing occurs, we might expect different detection thresholds for high frequency stimuli presented from different elevations, or, conversely, we might expect them to be the same. We will report data, collected in a soundfield, concerning the interaction, if any, of sound source elevation and auditory masked detection of high frequency stimuli.

2aPPb30. Spatial external noise and detection. M. Torben Pastore (College of Health Solutions, Arizona State Univ., 4 Irving Pl., Troy, New York 12180, m.torben.pastore@gmail.com) and William Yost (College of Health Solutions, Arizona State Univ., Tempe, AZ)

In vision, a change in target position can cause an otherwise stationary target to perceptually "pop out," thereby increasing its detectability. Previous work in this lab and others has shown that there is no corollary effect for auditory detection. The explanation most often given for this "null result" is that, while vision is encoded spatiotopically, audition is encoded tonotopically, and therefore changes in sound source position should not be expected to impact auditory detection in the way that target movement affects visual detection. This argument assumes that detection is largely a peripheral task. However, Green, Watson, and others have shown that auditory detection is affected by listener uncertainty about the frequency of presented target tones—this "external noise," in terms of frequency, suggests that detection involves processing at higher levels beyond the periphery (e.g., attention). It therefore stands to reason that auditory target stimuli may be less detectable when sound source location is randomized—a spatial manifestation of external noise. We will present data testing this hypothesis.

2aPPb31. Talker head orientation discrimination using only auditory cues. David L. Frazier (Speech and Hearing Sci., Univ. of Illinois, 1506 Pine Trace Ct, University Park, IL 60484, supermanfrazier@gmail.com) and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois, Champaign, IL)

Although many studies focus on human ability to localize a sound source, less is known about human ability to determine the physical orientation of a given sound source. In our study, we assessed listeners' ability to detect changes in talker head orientation. Participants with normal hearing were asked to detect head orientation changes (relative to 0 deg, *i.e*., directly facing the listener) for two male and two female talkers. We found listeners

are sensitive to changes of approximately 40 deg in talker head orientation using only auditory cues. This is less sensitive than what humans have displayed with only visual cues. These findings indicate that auditory cues are available for head orientation discrimination, which may be of greater utility when visual cues are unavailable.

2aPPb32. Measured localization performance at low stimulus levels with an adaptive tracking task. Nathaniel Spencer (Airforce Res. Lab., 2610 7th St., Area B, Bldg. 441, Wright-Patterson Airforce Base, OH, spencernj80@gmail.com), Eric R. Thompson, and Brian Simpson (Airforce Res. Lab., Wright-Patterson AFB, OH)

Sabin *et al*. (2005) measured localization accuracy for 250-ms broadband noises that varied in location along a spherical surface, and varied in stimulus level, from 0 to 60 dB above the detection threshold. They found that, for lower stimulus levels, responses tended to be biased towards lower elevations and to the front, and were more accurate at higher stimulus levels. Whereas Sabin *et al*. (2005) randomized level and location between trials, the current study fixed location and increased level until a correct response was given (or the 80 dB SPL limit was reached). Our initial level was 12.5 dB SPL, and our step size 2.5 dB SPL. Responses tended to be biased towards lower vertical elevations and to the front at low stimulus levels, and be more accurate at higher levels, consistent with Sabin *et al*. The percentage of front/back errors was generally greater for locations in the rear hemifield relative to those in the frontal hemifield, and for locations at higher elevations relative to locations at lower elevations. Audibility analysis, performed using a KEMAR manikin, showed that localization errors tended to decrease the most when audible bandwidth increased at the acoustically better ear.

2aPPb33. Updating of spatial selective auditory attention undercompensates for listener head movement. Ewan A. Macpherson, Miyoung Jeon, and Serena Ransom (National Ctr. for Audiol., Western Univ., 1201 Western Rd., Elborn College 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

Listeners can use spatial selective auditory attention (SSAA) to focus on one talker in a complex acoustic scene. In our dynamic SSAA task, listeners oscillate their heads ~±40 deg at ~0.5 Hz while five different simultaneous sequences of four spoken digits are presented from loudspeakers at 0 deg azimuth (the target) and ±22.5 deg and ±45 deg azimuth (four distractors); listeners report the target sequence heard. We have observed [ASA, Minneapolis 2018] that listeners are more likely to misreport distractors that are centrally located in head-centered coordinates at the moment of presentation, and that under static conditions, performance declines with increasing target eccentricity—suggesting either that listeners cannot rapidly update the focus of their SSAA to compensate head motion ("lag") or that they have difficulty directing SSAA eccentrically ("low gain"). To differentiate these alternatives, spatio-temporal maps of SSAA, conditioned on head orientation and direction of motion during each digit, were derived by computing the percentage of reported digits corresponding to those emitted by each loudspeaker. The spatial pattern of errors depended primarily on head orientation and not on direction of motion, suggesting that in this task SSAA tends to remain centrally focused in head-centered coordinates (low gain) rather than lagging dynamic head position.

2aPPb34. Development of duration discrimination during adolescence. Jennifer Gay, Merri Rosen (Dept. of Anatomy and Neurobiology, Northeast Ohio Medical Univ., Rootstown, OH), and Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., 1325 Theatre Dr., Kent, OH 44242, jhuyck@kent.edu)

Temporal processing, which is important for comprehending speech, matures over an extended developmental period. Here we investigated duration discrimination during adolescence. Listeners aged 8–19 years (four age groups) heard three broadband noises on each trial, and indicated which, if any, of the noises was different in length (longer or shorter). The broadband noises were 15, 30, 50, 100, or 200 ms in duration. During each of two sessions on consecutive days, each combination of stimulus durations (e.g., 15 vs. 100 ms) were presented a total of twenty times in pseudo-randomized order, including identical comparisons (e.g., 100 vs. 100 ms) to enable calculation of false alarm rates. All age groups showed a higher (better) sensitivity (d') for comparisons whose durations were more different from one another (e.g., 15 vs. 200 ms). Nevertheless, duration discrimination abilities continued to develop up to age ~14 years: 8- to 10-year-olds did not differ from 11- to 13-year-olds, but 11- to 13-year-olds had lower sensitivity than 14- to 17-year-olds and young adults, who themselves did not differ from one another. There was no learning between sessions and the interactions were not significant. Thus, even simple duration discrimination abilities may continue to develop into adolescence.

Session 2aSA

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, Noise, and Architectural Acoustics: Acoustic Metamaterials I

Christina J. Naify, Cochair
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Alexey S. Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd., West Bethesda, MD 20817

Invited Papers

8:00

2aSA1. Low frequency sound absorption by compliant Helmholtz resonators. Shichao Cui and Ryan L. Harne (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., Columbus, OH 43210, cui.408@osu.edu)

A traditional Helmholtz resonator is a rigid-walled cavity and an open neck, which result in one acoustic resonance. In this study, a soft-walled cavity with an open neck, a compliant Helmholtz resonator, is investigated for broadband and large absorption of sound at low frequencies. By applying the soft materials, the dynamics of the compliant walls and the material impedance are coupled to the acoustic pressure filed in the cavity. Thus the impedance of the compliant resonator system as seen by an incident acoustic wave is determined by the interaction of the wall dynamics, material properties, and acoustic domain. Compared to a traditional Helmholtz resonator with the same geometric dimensions, the compliant Helmholtz resonator exhibits multiple resonances with large absorption of sound. In addition, strategic design of the structural composition and the material selection leads to resonant behavior at frequencies lower than the conventional Helmholtz resonance. Through exploring an analytical model developed to characterize the resonator, the threshold on material and structural compliance to yield traditional Helmholtz behavior is also illuminated.

8:20

2aSA2. Evaluation of metamaterial unit cell analysis techniques. Amanda Hanford, Benjamin Beck, Aaron J. Stearns, and Andrew S. Wixom (Appl. Res. Lab, Pennsylvania State Univ., P.O. Box 30 - MS 3230D, State College, PA 16804, ald227@psu.edu)

The field of acoustic metamaterials has produced novel materials in a wide variety of applications. An important step in designing a metamaterial is unit cell analysis with subwavelength geometry. There are several techniques used for unit cell analysis when designing acoustic properties of interest for metamaterial applications. Such techniques include, but are not limited to, band diagrams, effective material properties, or half-space homogenization. This talk discusses the challenges and tradeoffs between analysis techniques and types of structures that lend to one method or another. Unit cell analysis methods will be evaluated to perform trade space exploration, including validation and parametric studies for metamaterial design.

8:40

2aSA3. Passive Non-Reciprocity in Asymmetrical, Hierarchical, Nonlinear Metamaterials. Michael J. Leamy, Amir Darabi, Lezheng Fang, Matthew Fronk (Georgia Tech, 771 Ferst Dr. NW, Atlanta, GA 30332-0405, michael.leamy@me.gatech.edu), and Alex Vakakis (Univ. of Illinois Urbana-Champaign, Urbana, IL)

Reciprocity is a property of linear, time-invariant systems whereby the energy transmission from a source to a receiver is unchanged after exchanging the positions of the source and receiver. Non-reciprocity, on the other hand, violates this property and can be introduced to systems if time-reversal symmetry and/or parity symmetry is lost, or by introducing nonlinearity. While many studies have induced non-reciprocity by active means, considerably less attention has been given to acoustical structures that passively break reciprocity. In this talk, we will discuss passive, strongly nonlinear periodic structures which exhibit giant reciprocity breaking under impulsive and/or harmonic excitation. Numerical means are employed to generate dispersion curves, as a function of wave energy, which differ for left-to-right from right-to-left propagation. These dispersion curves become reciprocal at the limiting cases of low and high energy, which can be shown analytically. In between, varying degrees of non-reciprocity can be achieved, and in some regimes, giant reciprocity breaking is achieved with very little harmonic distortion. This suggests many possibilities for passive, non-reciprocal devices that operate with near-linear behavior.

9:00

2aSA4. Contrast parameter characterizes Bragg frequency gap in periodic flexural beams. Thomas Gallot (Instituto de Física, Facultad de Ciencias, Universidad de la República, Igua 4225, Montevideo 11400, Uruguay, tgallot@fisica.edu.uy), Adrien Pelat, and Francois Gautier (Université du Mans, Laboratoire d'Acoustique de l'Université du Maine, Le Mans, France)

Periodic structures exhibit frequency bands where destructive interferences prohibit wave propagation. Such behavior can be useful to mitigate vibrations, in particular when structure lightening is important. Band gap properties can be deduced from Bloch's theory approach, Plane Wave Expansion or Multiple Scattering methods. These methods require a full description of the unit cell geometry and its mechanical properties. In this work we focus on an ideal unit cell geometry to highlight the role of a contrast parameter in the band gap opening process. We consider flexural waves in beams and demonstrate analytically that the contrast parameter fully controls the first Bragg band gap. Numerical simulation and experiments on a beam demonstrator proves that the gap bandwidth is independent of the section geometry: only the flexural rigidity is involved. We propose a semi-analytic model for the central frequency gap that depends on the mass distribution. The established algebraic expressions for the band gap bandwidth and central frequency successfully matches the results in the practical case and can be used to design flexural wave cut-band filters. Finally, symmetry considerations explains the experimental observation of a second band gap, also suitable for vibration mitigation, due to coupling of flexural and compressional waves.

9:15

2aSA5. Theoretical study of a metasurface-based sound absorber. Jun Ji (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., EBIII, Raleigh, NC 27695-7910, jji5@ncsu.edu), Ni Sui, Xiang Yan (Mech. and Aerosp. Eng., North Carolina State Univ., Providence, RI), Fuh-Gwo Yuan, and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

A metasurface-based sound absorber panel is proposed to achieve high sound absorption with a deep sub-wavelength thickness and a relatively broad bandwidth. The panel composed of different periodically-arranged Helmholtz resonators has been designed by a radiation impedance model which takes the mutual radiation impedance into account. Numerical simulations based on COMSOL Multiphysics show that the radiation impedance model yields more accurate results compared with the traditional equivalent surface impedance method. Broad bandwidth of sound absorption is achieved under proper coupling by carefully tuning the geometry and distance between Helmholtz resonators.

9:30

2aSA6. A Lagrangian formulation of metamaterial homogenization. Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil)

Many metamaterials may be well described as multi-scale systems where the large scale is represented by the wavelength and the small scales are represented by inhomogeneity lengths. The effective material properties of a metamaterial of this sort may be derived by considering the averaged behavior of material elements that are considered "infinitesimal" relative to the wavelength but is on the order of the inhomogeneity lengths. The averaging of the small-scale features enables a straightforward process to describe the metamaterial mechanics in terms of a Lagrangian formalism, and the macroscopic constitutive equations may be obtained by deriving the Euler-Lagrange equations for the macroscopic field quantities. This energetic approach to homogenization may be used to estimate both linear and nonlinear material properties. This talk describes this homogenization method for one-dimensional systems and compares the resulting effective material properties with other homogenization methods.

9:45–10:00 Break

10:00

2aSA7. Transverse to longitudinal mode conversion via nonlinear inertial effects in a locally resonant metamaterial plate. Samuel P. Wallen and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, walles@uw.edu)

Acoustic and elastic metamaterials have potential for tailoring wave propagation in ways beyond the capabilities of conventional materials and have been shown to exhibit a myriad of rich dynamical effects, such as negative effective properties, cloaking, extreme damping, and non-reciprocity. However, most of the existing literature has been focused on linear dynamics in which these behaviors are restricted to narrow frequency bands. More recently, nonlinearity has been investigated as a method of increasing this bandwidth, as well as a means to broaden the palette of available dynamics. Past studies on nonlinear acoustic and elastic metamaterials have demonstrated effects not easily accessible in the linear regime, including harmonic generation, mode hopping and conversion, tunable band gaps, chaos, and intrinsic localized modes. In this work, we present a model for a nonlinear elastic plate metamaterial that exhibits mode conversion between transverse and longitudinal waves. The mode conversion is enabled by cantilevered beam attachments undergoing large-angle out-of-plane oscillations. By analyzing the unit cell of this locally-resonant plate, we interpret the transverse-longitudinal coupling as a nonlinear, anisotropic effective mass density. The mode conversion effect is demonstrated via direct numerical simulations. [Work supported by NSF and ARL:UT.]

10:15

2aSA8. Ultra-open acoustic metamaterial silencer. Reza Ghaffarivardavagh, Jacob Nikolajczyk (Mech. Eng., Boston Univ., 8 Saint Mary St., Photonic Bldg., Rm. 832, Boston, MA 02215, ghaffari@bu.edu), Stephan Anderson (Boston Univ. Medical Ctr., Boston, MA), and Xin Zhang (Mech. Eng., Boston Univ., Boston, MA)

Recently, with advances in acoustic metamaterial science, the possibility of sound attenuation using subwavelength structures, while maintaining permeability to air, has been demonstrated. However, the ongoing challenge addressed herein is the fact that among such air-permeable structures to date, the open areas represents only small fraction of the overall area of the material. In the presented work, in order to address this challenge, we firstly demonstrate that a transversely-placed bilayer medium with large degrees of contrast in the layers' acoustic properties exhibits an asymmetric transmission, similar to the Fano-like interference phenomenon. Next, we utilize this design methodology and propose a deep-subwavelength acoustic metasurface unit cell comprising nearly 60% open area for air passage, while serving as a high-performance selective sound silencer. Finally, the proposed unit cell performance is validated experimentally, demonstrating a reduction in the transmitted acoustic energy of up to 94%. This ultra-open metamaterial (UOM) design, leveraging a Fano-like interference, enables high-performance sound silencing in a design featuring a large degree of open area, which may find utility in applications in which highly efficient, air-permeable sound silencers are required, such as smart sound barriers, fan or engine noise reduction, among others.

10:30

2aSA9. Design, optimization, and fabrication of mechanical metamaterials for elastic wave control. Timothy F. Walsh (Computational Solid Mech. and Structural Dyn., Sandia National Labs., P.O. Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov)

Harsh shock and vibration environments are commonly encountered in engineering applications involving dynamic loading. Acoustic/elastic metamaterials are showing significant potential as candidates for controlling wave propagation and isolating sensitive structural components. However, these materials have complex microstructures that must be properly designed to achieve their desired properties. In this talk we will present strategies for PDE-constrained design optimization of locally resonant elastic/acoustic metamaterials. We will present a variety of resonator geometries that can be easily optimized for wave control applications, along with fabrication details involving multi-material additive manufacturing. A variety of objective functions will be compared for their effectiveness in designing mechanical filters. Numerical examples will be presented for vibration isolation and acoustic cloaking applications. Sandia National Laboratories is a multimission laboratory managed and operated by National Technology and Engineering Solutions of Sandia, LLC, a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Departmentof Energy's National Nuclear Security Administration. With main facilities in Albuquerque, NM, and Livermore, CA, Sandia has major R&D responsibilities in national security, energy and environmental technologies, and economic competitiveness.

10:45

2aSA10. Measuring anisotropy in underwater inertial metamaterials. Colby W. Cushing, Preston S. Wilson, Michael R. Haberman (Appl. Res. Labs and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, colby.cushing@utexas.edu), Chen Shen, and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

The acoustic behavior of acoustic metafluids, designed and fabricated for underwater applications has been studied experimentally. Unit cells consist of elastic elements coupled via compliant layers to produce negligible shear modulus and anisotropic dynamic effective density and phase speed in orthogonal directions. Finite element simulations were used to design a unit cell that exhibits subsonic phase speeds in one direction and supersonic speeds in the orthogonal direction. Numerous samples were constructed to experimentally validate these predictions. Various combinations of materials were employed to enhance anisotropy, simplify construction, and reduce unwanted effects. Uniformly-spaced samples were tested from 0.05-10 kHz in a one-dimensional, resonator tube filled with degassed water. An electrodynamic shaker excited the system with frequency-modulated chirps. The system response was recorded using a hydrophone positioned near the top of the tube, and the resonance frequencies were used to infer the phase speeds for each mode. To extract material properties from these measurements, an effective medium model was used to represent the material-water mixture and an exact analytical dispersion relation was used to correct for elastic waveguide dispersion. Extracted material properties indicated anisotropy was achieved, and were found to be in good agreement with the as-designed properties. [Work supported by ONR.]

11:00

2aSA11. Experimental demonstration of an underwater acoustic leaky wave antenna. Michael Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 1, Austin, TX 78758, mikelee@arlut.utexas.edu), Curtis Wiederhold (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Kyle Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Christina J. Naify (none, Sacramento, CA), and Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Low-power acoustic imaging instrumentation can have a significant impact on underwater exploration and monitoring by enabling longer duration missions. Acoustic leaky wave antennas (LWAs), which consist of a dispersive analogue aperture coupled to a single electro-mechanical transducer, are a promising technological solution to address the demanding requirements of low-power underwater imaging. The dispersive nature of an LWA permits steering an acoustic beam from forward-to-backward directions by simply changing the frequency of the electro-mechanical transducer when used as either an acoustic source or receiver. While acoustic LWAs have been shown to achieve imaging resolution on par with active phased arrays, to date they have only been demonstrated for air-borne acoustic waves. This work presents the first realization of an underwater acoustic LWA using a leaky elastic waveguide consisting of a finite array of locally resonant elastic structures coupled to a single piezoelectric transducer. Finite element analysis was used as a design tool, and preliminary measurements in an underwater testing facility demonstrate forward-to-backward radiated beam scanning of a periodic, heterogeneous aluminum antenna. [Work supported by the Strategic University Research Partnership at NASA-JPL.]

11:15

2aSA12. Measurement of additively manufactured anisotropic three-dimensional underwater pentamode materials. Colby W. Cushing, Preston S. Wilson, Michael R. Haberman (Appl. Res. Labs and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, colby.cushing@utexas.edu), Xiaoshi Su, and Andrew Norris (Dept. of Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

Pentamode (PM) materials are three-dimensional (3D) elastic lattice materials that can be designed to match the acoustic impedance of water while also minimizing shear modulus of the sample over wide frequency ranges. Further, the lattice structure provides the degrees of freedom necessary to produce strongly anisotropic stiffness that is useful for transformation acoustics [Su *et al*., *J. Acoust. Soc. Am.* **141**(6) (2017)]. In the current work, anisotropic sub-wavelength 3D metallic PM samples were fabricated using additive manufacturing for measurement of their effective material properties. The samples were distributed uniformly in a one-dimensional, water-filled resonator tube. An electrodynamic shaker excited the system and the system response was recorded using a hydrophone positioned near the top of the tube. The resulting resonance frequencies were then used to infer the phase speeds for each mode in the fluid-filled elastic waveguide. Two effective medium models were used to infer the PM-water mixture properties: (i) Wood-Mallock mixture law and (ii) a self-consistent micromechanical model that explicitly considers material anisotropy. Experimental results are compared with simulations and observations are drawn on material property extraction using the resonance tube technique when elastic anisotropy is present. [Work supported by ONR.]

Session 2aSCa

Speech Communication: Perception of Speech Directed Toward Infants and Children

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Chair's Introduction—8:00

Invited Papers

8:05

2aSCa1. ManyBabies1: Infants' preference for infant-directed speech. Melanie Soderstrom (Psych., Univ. of Manitoba, 190 Dysart Rd., Winnipeg, MB R3T 2N2, Canada, m_soderstrom@umanitoba.ca)

There is growing concern about the replicability of basic findings in psychology, including in infancy research (Frank *et al*., 2017). ManyBabies is a large-scale international collaboration to replicate basic empirical findings in infancy. Our first project is ManyBabies1, which examines infant preference for infant-directed speech (IDS). Over 70 laboratories collaborated to collect data from over 2500 infants aged 3–15 months. Stimuli consisted of speech produced in a semi-naturalistic elicitation task where fifteen different mothers who spoke North American English talked about a series of novel and familiar objects to their infant and separately to an experimenter. A set of 8 passages each in IDS and adult-directed speech (ADS) were created after a comprehensive norming process. Infants were tested in 3 primary methods: eyetracking, central fixation and headturn preference. Overall, the effect of preference for IDS was replicated, although the calculated effect was smaller than that reported by meta-analysis (Dunst *et al*., 2012, metalab.stanford.edu). Preference for IDS increased across development. The success of this project shows that infant research conducted collaboratively and at-scale can answer new questions about the replicability and generalizability of infant findings across different laboratory, methodological, and infant participant characteristics.

8:25

2aSCa2. ManyBabies1 part 2: Influences of language experience on infant-directed speech preference. Melanie Soderstrom (Psych., Univ. of ManitobaMB, 190 Dysart Rd., Winnipeg, MB R3T 2N2, Canada, m_soderstrom@umanitoba.ca) and Krista Byers-Heinlein (Psych., Concordia Univ., Montreal, QC, Canada)

ManyBabies1, our first effort at a large scale collaborative infant experimental study, provided a conceptual replication of the well-known phenomenon of infant preference for the characteristics of Infant-directed speech (IDS). One important question that has largely been unanswered by extant literature is how much the IDS preference is dependent on experience with a specific language. How do infants respond to IDS that is in a non-native variety, and how does their listening affect this preference? ManyBabies 1 used a consistent stimulus set of North-American English (NAE), which allowed us to answer this questions using two approaches. First, because participating ManyBabies 1 labs were located around the world, we were able to compare monolingual infants from a range of native-language backgrounds. We found that the preference for North American English IDS was larger for infants whose native language was NAE than for infants who had a different native language. Second, we conducted a sister project, ManyBabies 1 Bilingual, which tested infants from a variety of bilingual backgrounds. Bilinguals have similar total language experience and maturation as monolinguals, but their experience is divided across two or more languages. Planned analyses will examine monolingual-bilingual differences, and "dose-response" effects of exposure to NAE.

8:45

2aSCa3. Infant-directed speech facilitates neural encoding of speech during infants' first year of life. Marina Kalashnikova (The Basque Ctr. on Cognition, Brain and Lang., Paseo Mikeletegi 69, San Sebastian, Guipuzkoa 20009, Spain, m.kalashnikova@bcbl.eu)

Infant-directed speech (IDS) is the special speech register that parents use when addressing young infants. Compared to adult-directed speech (ADS), it is characterised by positive affect, high pitch and wide pitch range, and acoustically exaggerated speech sounds. These acoustic components of IDS have been proposed to facilitate early language acquisition by attracting infants' attention to speech, or by providing them with a type of speech input that is easier to perceive and learn, or both. Indeed, infants are more successful in a variety of behavioural language processing tasks when presented with IDS stimuli, but it is unclear whether these effects are due to the linguistic function of IDS or solely to its attention-grabbing qualities. I will present three studies that address this question by directly

measuring infants' neurophysiological responses to IDS and ADS. Our findings demonstrate that IDS elicits greater neural activation and more mature neural response patterns compared to ADS in measures of neural entrainment to continuous speech, phonemic discrimination, and semantic processing in 4-to-9-month-old infants. This evidence indicates that beyond capturing infants' attention to speech, IDShas a privileged status in facilitating neural encoding of speech, which may augment infants' early speech processing and even later language development.

9:05

2aSCa4. Effects of hearing loss and amplification device on infants' perception of infant-directed speech. Tonya Bergeson (Commun. Sci. and Disord., Butler Univ., 4600 Sunset Ave., JH246, Indianapolis, IN 46208, tbergeso@butler.edu), Yuanyuan Wang, and Derek Houston (Otolaryngol., The Ohio State Univ., Columbus, OH)

Caregivers typically speak to their infants in a speech register known as motherese or infant-directed speech (IDS; Fernald and Simon, 1984; Snow, 1977). Infants also prefer to listen to IDS over adult-directed speech (ADS) (Cooper and Aslin, 1990; Fernald, 1985; Singh *et al*.). Moreover, mothers adjust their IDS to factors such as infant age (e.g., Kitamura and Burnham, 2003; Newman and Hussain, 2006). Another factor that could potentially affect mother-infant interactions is infant hearing loss. Studies have shown that mothers adjust their speech based on infants' hearing experience (e.g., Bergeson *et al*., 2006; Kondaurova and Bergeson, 2011; Wieland *et al*., 2015). Do infants with hearing loss pay attention to IDS? Two recent studies suggest that infants who use hearing aids or cochlear implants demonstrate increased attention to IDS over both ADS and silent trials (Wang*et al*., 2017; 2018). One feature that might regulate infant attention, namely target word repetition, is correlated with vocabulary skills in later childhood (Wang *et al*., in preparation). In this talk, we will address infants' attention to speech and the role IDS serves in both cognitive-social and linguistic development for typically hearing infants and those with hearing loss.

9:25

2aSCa5. Daylong acoustic recordings of family and child speech using the HomeBank database. Mark VanDam (Speech & Hearing Sci., Washington State Univ., P.O. Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Paul De Palma (Comput. Sci., Gonzaga Univ., Spokane, WA), Melanie Soderstrom (Psych., Univ. of Maintoba, Winnipeg, MB, Canada), Marisa Casillas (Max Planck Inst. for PsychoLinguist, Nijmegen, The Netherlands), Alex Cristia (Laboratoire de Sci. Cognitives et Psycholinguistique, Département d'études Cognitives, PSL Res. Univ., Paris, France), Elika Bergelson (Psych. & Neurosci., Duke Univ., Durham, NC), Anne Warlaumont (Commun., Univ. of California, Los Angeles, Los Angeles, CA), Daniel Olds (Comput. Sci., Washington State Univ., Spokane, WA), and Brian MacWhinney (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

HomeBank (https://homebank.talkbank.org/) is an online database of multi-hour, naturalistic audio recordings of child and family everyday experiences. Corpora in the database include (1) raw audio recordings; (2) corpus metadata including for example social details, standardized test scores, disability reports and status, family data such as number and quantity of siblings, orthographic transcriptions, output of diarization or automatic-speech recognition processing; and (3) tools for analyzing the data (https://github.com/homebankcode/) in a variety of domains. There are currently over a dozen corpora representing over 1100 multi-hour (often daylong) audio recordings. Use of the database is increasing in the scientific community, including researchers of speech, language, computer science, digital signal processing, automatic speech recognition, health sciences, and human development. The utility and extensibility of HomeBank is demonstrated in this talk with several current and ongoing projects that make critical use of the data. We discuss diarization and automatic speech processing techniques, speech and language use in pre-industrial families and groups, early perception and processing in infants, family speech and language dynamics in families of children with disorders, and database management. We are actively soliciting both users and contributors to the database.

Session 2aSCb

Speech Communication and Psychological and Physiological Acoustics: Acoustic Phonetic Properties of Speech Directed Toward Infants and Children

Mark VanDam, Cochair
Speech & Hearing Sciences, Washington State University, P.O. Box 1495, Spokane, WA 99202

Laura Dilley, Cochair
Department of Communicative Sciences, Michigan State University, East Lansing, MI 48824

Chair's Introduction—10:15

Invited Papers

10:20

2aSCb1. Understanding acoustic-phonetic environments of prelingual children with cochlear implants: Challenges, tools, and insights. Laura Dilley (Dept. of Communicative Sci., Michigan State Univ., East Lansing, MI 48824, ldilley@msu.edu)

Variability in auditory and linguistic environments experienced by prelingual children with cochlear implants (CIs) potentially helps explain variation in their language outcomes. Here, results are presented from several studies from our lab investigating early acoustic-phonetic environments of children with CIs. One collaborative study conducted over 10 years investigated individual differences in acoustic-phonetic quality of maternal speech as predictors of language outcomes in children with CIs. Results showed that properties of maternal speech recorded in the lab—as indexed in part by measures derived from vowel formants and fundamental frequency—significantly predicted infants' language outcomes and growth on multiple standardized assessments two years after cochlear implantation. Other work has examined the usefulness of a widely adopted commercial automatic speech processing technology—the Language ENvironment Analysis (LENA) system—for investigating individual differences in auditory environments. Results from our study suggested considerable variability across samples in LENA's accuracy at identifying adult speech, limiting LENA's value for investigating individual differences in acoustic-phonetic input to children with CIs. Finally, an update is provided on current work using combined signal processing and hand-coding approaches aimed at investigating variability in acoustic input to children with CIs in their home environments. [Work supported by NIH grant R01DC008581.]

10:40

2aSCb2. Using naturalistic paradigms to study how adult speakers accommodate infant listeners' unique processing demands. Elise A. Piazza, Marius Catalin Lordan, Liat Hasenfratz, Uri Hasson, and Casey Lew-Williams (Princeton Univ., 238E PNI, Princeton, NJ 08540, epiazza@princeton.edu)

Communication is inherently social and requires an efficient exchange of complex acoustic cues between individuals. What are the behavioral and neural processes that allow young listeners to understand, couple to, and learn from adult speakers in complex, everyday interactions? In one study, we recorded mothers' natural speech during play and reading and uncovered a pervasive timbre fingerprint of infant-directed speech (IDS) that generalized across 10 diverse languages. Classification of IDS and adult-directed speech (ADS) was driven by a statistical summary measure that concisely describes the vocal spectrum and could not be explained by pitch alone. In a second study, using dual-brain functional near-infrared spectroscopy (fNIRS), we measured the real-time neural dynamics of communication between infants and adults during natural interaction. We found that the infant prefrontal cortex (PFC) tracked several communicative cues, including the adult's pitch variability, with high temporal precision. Furthermore, infant-adult neural coupling was significantly greater when the members of a dyad interacted with each other than when they performed control tasks. Surprisingly, PFC activation in the infant brain slightly preceded similar activation in the adult brain, which crucially advances our understanding of children's influence over the accommodative behaviors of the caregivers around them during everyday communication.

11:00

2aSCb3. Child-directed speech in noise: Listener- and environment-related changes in speech acoustics. Nicholas A. Smith (Dept. of Speech, Lang. and Hearing Sci., Univ. of Missouri, Columbia, MO 65211, smithnich@health.missouri.edu), Christine A. Hammans, Timothy J. Vallier (Boys Town National Res. Hospital, Omaha, NE), and Bob McMurray (Dept. of Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Talkers adapt their speech in various ways according to the demands of their listeners and the communicative context. Mothers and their preschool children participated in a real-time interactive speech production/perception paradigm, in which mothers instructed their children (or an adult listener), to select the picture corresponding to the target word. The task was performed at low and high levels of

background noise (56 and 76 dB SPL, delivered through headphones), to examine the effects of decreased audibility on speech production. Acoustic-phonetic analyses of child-directed speech (CDS) and adult-directed speech (ADS) productions of target words and carrier phrase (e.g., "Find pig"), revealed that mothers significantly enhanced the suprasegmental properties (i.e., pitch, intensity, and duration) of target words in CDS and at higher noise levels, but provided limited evidence for the hyperarticulation of the segmental properties of speech (i.e., formant frequencies of vowels, or voice-onset times of stop consonants). Results suggest that while some aspects of articulatory control are readily amenable to change as function of task/listener demands, others may not be. Understanding these capacities and constraints in the talking caregiver is relevant to theories of hyperarticulation in infant-directed speech.

11:20

2aSCb4. Temporal coordination of vocal turn taking between mothers and their children with hearing loss. Maria V. Kondaurova (Dept. of Psychol. & Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 46292, maria.kondaurova@louisville.edu), Nicholas A. Smith (Dept. of Speech, Lang. and Hearing Sci., Univ. of Missouri – Columbia, Columbia, MO), Qi Zheng (Dept. of Biostatistics, Univ. of Louisville, Louisville, KY), Jessa Reed (Dept. of Otolaryngology-Head and Neck Surgery, The Ohio State Univ. Medical Ctr., Columbus, OH), and Mary K. Fagan (Dept. of Commun. Sci. and Disord., Crean College of Health and Behavioral Sci., Chapman Univ., Orange, CA)

Normal-hearing (NH) infants participate in social exchanges soon after birth. What does vocal turn-taking look like in children with hearing loss after cochlear implantation? The study examined the prevalence and temporal structure of vocal turns during spontaneous interactions between mothers and their children with cochlear implants (CIs) over the first year after implantation compared to interactions between mothers and children with normal hearing. Mothers' play with children with CIs (n = 12) were recorded at 3 (mean age 18.3 mo) and 9 (mean age 27.5 mo) months post CI. Mothers with age-matched hearing children (n = 12) were recorded at the corresponding time points. The CI group initially differed from the NH group in several ways (i.e., fewer vocal turns, more simultaneous speech, and longer between-speaker pauses) but progressed to NH levels by 9 months post CI, demonstrating the positive effects of CIs. Dyadic effects were also observed in the timing of mothers' responses, which were related to those of their children. However, children with CIs continued to show an atypical pattern in the relative timing of between- vs within-speaker pauses across both test sessions, indicating a potentially protracted time course for the influence of CIs on dyadic interactions.

11:40

2aSCb5. Developmental cascades in reciprocal vocal signaling between infant and caregiver in typical development and autism. Gordon Ramsay, Shweta Ghai, Mitra Kumareswaran, Morgan Edwards (Dept. of Pediatrics, Emory Univ. School of Medicine, Marcus Autism Ctr., 1920 Briarcliff Rd. NE, Atlanta, GA 30329, gordon.ramsay@emory.edu), and Jhonelle Bailey (Dept. of Psych., Univ. of Miami, Coral Gables, FL)

Adults habitually adopt a special vocal register when talking to children. Current evidence suggests that the acoustic properties characterizing infant-directed speech change over development, in response to changes in infant vocal behavior, but the factors driving these changes remain largely unknown. The goal of this study is to elucidate developmental progressions in the acoustic structure of infant-directed speech over the first two years of life, and to determine the origin of these changes in infant vocal response. Samples of adult- and infant-directed speech and infant vocalizations were extracted from audio recordings of 10 typically developing infants and 10 infants later diagnosed with autism and their mothers, collected from 0 to 24 months using LENA technology. Multitaper analysis was used to determine the time-varying harmonic structure of each utterance, deriving indices summarizing differences in source and filter properties between infant- and adult-directed speech. In typical development, transitions were found towards the end of the first year of life from baby talk, emphasizing prosodic properties, towards mature child-directed register, emphasizing resonance dynamics. In autism, caregivers persisted in baby talk or shifted into adult-directed register, concurrent with disruptions of vocal contingency, which may be stimulating the transition.

Session 2aSPa

Signal Processing in Acoustics: Beamforming, Detection and Localization

Kathleen E. Wage, Chair
George Mason University, 4400 University Drive, MSN 1G5, Fairfax, VA 22030

Contributed Papers

8:00

2aSPa1. Performance weighted blending of nested array processors. Jeff Tucker (ECE, George Mason, 1511 Black Eyed Susan Ln., Vienna, VA 22182, jtucke16@gmu.edu) and Kathleen E. Wage (ECE, George Mason, Fairfax, VA)

Nested arrays consist of two uniform subarrays: a short aperture array with half-wavelength spacing and a long aperture array with greater than half-wavelength spacing. Two approaches for estimating the scanned response require computing either the product or the minimum of the subarray responses in each look direction. In multi-source environments, the multiplicative and min scanned responses may be corrupted by cross terms [Wage, *Acoustics Today* (2018)]. When the sources are uncorrelated, snapshot averaging is often used to mitigate cross term interference. The min processor's response can contain cross terms that do not decay with snapshot averaging. Alternatively, the multiplicative processor's response contains cross terms that decay, though large numbers of snapshots may be required for the cross terms to fall below the level of those in the min processor. This talk proposes a performance weighted combination of the two processors based on Buck and Singer's (IEEE, 2018) blended dominant mode rejection beamformer. The same performance weighting function can be used to combine the nested multiplicative and min processors. The resulting universal processor adjusts the weighting in each look direction as the number of snapshots increases to favor the multiplicative processor as its cross terms average out. [Work supported by ONR]

8:15

2aSPa2. Post-processing for underwater active detection and experimental results. Lin Ma, Anbang Zhao, and Chunsha Ge (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Harbin, HLJ 150001, China, malin@hrbeu.edu.cn)

In practical engineering, the target echo signals are commonly corrupted by noise and strong interferences that severely degrade the active detection performance. In the presence of noise and interferences in the detection results, the target trajectory cannot be easily recognized. A post-processing method is proposed to cancel the noise and interferences and to make the target trajectory better to be characterized. In accordance with the difference between noise and signal, a width discriminator is used to filter out some fake peaks appeared in the detection output. Considering the environmental effects, the format-based method is used to reduce the stationary interferences, for instance, reverberation due to rocks on the bottom. The post-processing method is employed and investigated in a target underwater detection experiment. The experimental results indicate that the post-processing is effective in active detection for an engineering application.

8:30

2aSPa3. Use of frequency-difference beamforming for scatterer localization. Alexander S. Douglass, Kely J. Markley, and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu)

Acoustic fields interacting with discontinuities or environmental variations lead to secondary scattered fields superimposed onto incident fields. As a result, the signal measured at a remote receiving array will be modified by the scatterer's presence. Prior work has shown that the effects of strong random scattering are reduced using frequency-difference beamforming to downshift the analysis to a below-band frequency known to be less impacted by scattering. In some applications, information about a scatterer is desired, such as location, shape, or composition. However, a relatively weak scattered field from a single scatterer is difficult to detect when the incident field dominates the measurements, as both conventional and frequency-difference beamforming ambiguity surfaces primarily provide the incident field's source information. Here, a subtraction-based algorithm is implemented with frequency-difference beamforming to locate the position of a single scatterer near a source. Simulations and water tank experiments with a 110 kHz center frequency signal are considered for incident and scattered fields with a 4 cm diameter spherical scatterer, where the scattered field's received energy is roughly 2% that of the incident field. The ability to downshift the frequency demonstrates a more robust implementation of subtraction-based algorithms for locating weak scatterers without prior location knowledge. [Work supported by ONR.]

8:45

2aSPa4. Application of adaptive line spectrum enhancer in adaptive matched filter. Xiaoliang Zhang (Systems Eng. Res. Inst. of CSSC, Beijing, China), Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin Eng. University, Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn), JiangQiao Li, Huangpu Li (Systems Eng. Res. Inst. of CSSC, Beijing, China), and Xianzhong Bu (Underwater Acoust., Harbin Eng. Univ., Harbin, China)

The most classical detector of active sonar and radar is the matched filter (MF), which is the optimal processor under ideal conditions. Aiming at the problem of active sonar detection, we propose a frequency-domain adaptive matched filter (FDAMF) with the use of a frequency-domain adaptive line enhancer (ALE). The FDAMF is an improved MF. In the simulations in this paper, the signal to noise ratio (SNR) gain of the FDAMF is about 18.6 dB higher than that of the classical MF when the input SNR is -10 dB. In order to improve the performance of the FDAMF with a low input SNR, we propose a pre-processing method, which is called frequency-domain time

reversal convolution and interference suppression (TRC-IS). Compared with the classical MF, the FDAMF combined with the TRC-IS method obtains higher SNR gain, a lower detection threshold, and a better receiver operating characteristic (ROC) in the simulations in this paper. The simulation results show that the FDAMF has higher processing gain and better detection performance than the classical MF under ideal conditions. The experimental results indicate that the FDAMF does improve the performance of the MF, and can adapt to actual interference in a way. In addition, the TRC-IS preprocessing method works well in an actual noisy ocean environment.

9:00–9:15 Break

9:15

2aSPa5. Statistical characterization of cross terms in snapshot-averaged multiplicative processors. Vaibhav Chavali and Kathleen E. Wage (Elec. Eng., George Mason Univ., 4217 University Dr., Fairfax, VA 22030, vchavali@gmu.edu)

Multiplicative processors generate spatial spectrum estimates for sparse arrays, such as nested and coprime arrays, by multiplying the beamformed outputs of two interleaved subarrays. Nested and coprime arrays achieve significant sensor savings compared to dense Uniform Linear Arrays (ULAs) since one or both subarrays are undersampled. Chavali *et al*. show that careful design of the subarrays and proper selection of beamformer weights can guarantee power pattern performance (response to single planewave source) comparable to a conventional ULA processor [*JASA* (2018)]. The multiplicative processors' response to multiple sources contains cross terms that appear as erroneous sources in the spectral estimate. The height of cross term peaks in the spectrum depends on the subarray design, beamformer weights, and signal powers. Ksienski and Pedinoff show that averaging the multiplicative processor output over snapshots reduces the power of uncorrelated cross terms [*IEEE* (1962)], though they do not explore how much averaging is required. This talk derives the statistics of the averaged cross terms assuming complex Gaussian planewave signals and noise. The statistical model is used to predict the number of snapshots required to eliminate uncorrelated cross term peaks from the multiplicative spectra. Analytical predictions show excellent agreement with planewave simulations. [Work supported by ONR.]

9:30

2aSPa6. Nested sensor array extension factors required to match the peak sidelobe height of a uniform linear array. Hossam Elsaadawy, Kayla M. Houte, Camille LeBlanc, James M. Slezak, and Kaushallya Adhikari (Elec. Eng., Louisiana Tech Univ., 501 Don Raneau Dr. #10348, Ruston, LA 71272, kmh074@latech.edu)

Nested sensor arrays (NSAs) are a type of non-uniform linear array (NULA) that reduce the total number of sensors used for a given aperture and retain the same resolution of a uniform linear array (ULA) [Pal and Vaidyanathan, 2010]. The peak sidelobe (PSL) height fails to fall below the acceptable standard of -13 dB associated with ULAs for basic NSA configurations. The basic NSA can be extended by a factor to improve the PSL height. Coprime sensor arrays, a related NULA design, have established sensor extension factors that have been found to reduce the PSL height to this -13 dB standard [Adhikari *et al*., 2014]. For NSAs, such an extension factor has not been established. This research finds the parameters required to extend an NSA to adequately reduce the PSLs. NSAs can significantly reduce the number of sensors required to yield a satisfactory beampattern, as low as twenty-five percent of the total aperture for some cases. Considering various array apertures, certain configurations successfully use a distinctly small aperture but require a higher percentage of those to be active. To verify our results, we construct a sixty-four sensor microphone array with which we test the optimal configurations. [Work supported by Louisiana Tech University.]

9:45

2aSPa7. Multiplicity of coprime pairs for extension of coprime sensor arrays. Pablo Johnson, Daniel Sartori, Tyler Trosclair (Elec. Eng., Louisiana Tech Univ., P.O. Box 5301, Ruston, LA 71272, pkj006@latech.edu), John Willis (Elec. Eng., Louisiana Tech Univ., Saint Francisville, LA), and Kaushallya Adhikari (Elec. Eng., Louisiana Tech Univ., Ruston, LA)

This research finds the optimum period of a basic coprime sensor array (CSA) needed to reduce the PSL height to -13 dB for both minimum and product processing methods. Our design does not have the constraint that the subarray lengths be one integer apart unlike in [Adhikari *et al*., 2014]. We apply MUSIC algorithm to the optimized CSA and compare the PSD estimate obtained by Vaidyanathan and Pal's covariance matrix estimation using a single sensor pair versus our covariance matrix estimated using all available sensor pairs. We show that minimum processing requires fewer periods than product processing to reach the PSL height of -13 dB. Larger coprime pairs require fewer periods and cause an increase in variance and a decrease in the number of lags available between the two subarrays. This decrease lowers the accuracy of the covariance matrix estimation. However, using all available sensor pairs to generate the covariance matrix greatly increases the accuracy compared to using a single sensor pair. When only one signal is present, both covariance estimates produce a similar, accurate DOA, but as the number of signals increases, our covariance estimate performs better. [Work supported by Louisiana Tech University.]

10:00

2aSPa8. Improving autonomous vehicle safety—Using acoustic source localization to influence the decision making capabilities of an autonomous vehicle. Digno Iglesias, Anthony Matriss, Akin Tatoglu, and Eoin A. King (Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu)

The manner in which autonomous vehicles (AVs) and other vehicles will coexist, and communicate with one another, is still unclear, especially during the prolonged period of mixed vehicles sharing the road. During this time, it will be important that AVs are equipped to respond to acoustic alerts in urban environments. Consider the case of an emergency vehicle in a city. In this situation, a visual perception system alone will not suffice, as it requires a direct view of the source. The goal of this ongoing project is to define a new set of audio-visual detection and localization tools to identify the location of an unexpected rapidly approaching emergency vehicle. In particular, this paper focuses on the localization of a sound source in a complex environment by combining a ray tracing approach with a direction-of-arrival algorithm. This algorithm reports a number of source directions, arising from multiple reflections in the environment. Results are combined with a three-dimensional map, acquired live from a mobile robot equipped with digital sensors including LiDAR, and a reverse ray tracing approach is used to triangulate the likely position of the source.

10:15

2aSPa9. Distributed sensing for acoustic source localization in indoor reverberant environments. Angela Bertolino, Pratik Gandhi, Arielle Joasil, Chester Obi, Kavitha Chandra, and Charles Thompson (ECE, UMASS, CACT fa203, Lowell, MA 01854, angela_bertolino@student.uml.edu)

The acoustic source localization problem relies on the estimate of the difference in time-delays of the signal received between a pair of acoustic sensors separated by a fixed distance. In the ideal case of a room with no reflections, one can identify the intersection of constant power level maps generated by multiple pairs of sensors to isolate the location of the source. The presence of reverberation however produces additional potential source locations that create uncertainty in the source location. In this research, impulse responses are simulated for a rectangular room using an image source model that incorporates frequency dependent absorption coefficients. These spatio-temporal impulse responses are applied in conjunction with room noise to simulate the signals recorded across a two-dimensional distribution of sensor pairs. The number of sensor-pairs and their spatial distribution that can optimally predict the source location in the presence of reverberant features is discussed.

Meeting of the Standards Committee Plenary Group
to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:

ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 43/SC 3, Underwater acoustics,
ISO/TC 108, Mechanical vibration, shock, and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machine systems,
and IEC/TC 29, Electroacoustics

R. D. Hellweg, Chair, P. D. Schomer, Vice Chair, U.S. Technical Advisory Group for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise
Hellweg Acoustics, 13 Pine Tree Road, Wellesley MA 02482
Schomer and Associates, 2117 Robert Drive, Champaign, IL 61821

R. W. Fischer, Chair, U.S. Technical Advisory Group for ISO/TC 43/SC 3 Underwater acoustics
Noise Control Engineering, Inc., 799 Middlesex Turnpike, Billerica, MA 01821

W. Madigosky, Chair of the U.S. Technical Advisory Group for ISO/TC 108 Mechanical vibration, shock, and condition monitoring
MTECH, 10754 Kinloch Road, Silver Spring, MD 20903

M. L'vov, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures
Siemens Energy, Inc., 5101 Westinghouse Blvd., Charlotte, NC 28273

D. D. Reynolds, Chair, U.S. Technical Advisory Group for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Court, Las Vegas, NV 89120

D. J. Vendittis, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machine systems
701 Northeast Harbour Terrace, Boca Raton, FL 33431

C. Walber, U.S. Technical Advisor for IEC/TC 29, Electroacoustics
diagnostics of machine systems
PCB Piezotronics, Inc., 3425 Walden Avenue, Depew, NY 14043 2495

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting

The meeting of the Standards Committee Plenary Group will follow the meeting of Accredited Standards Committee S2, which will be held on Monday, 13 May 2019, from 5:00 p.m. to 6:15 p.m.

The Standards Committee Plenary Group meeting will precede the meetings of the Accredited Standards Committees S1, S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:

Tuesday, 14 May 2019	11:00 a.m.–12:15 p.m.	ASC S1, Acoustics
Tuesday, 14 May 2019	2:00 p.m.–3:15 p.m.	ASC S3, Bioacoustics
Tuesday, 14 May 2019	3:30 p.m.–4:45 p.m.	ASC S3/SC1, Animal Bioacoustics
Tuesday, 14 May 2019	5:00 p.m.–6:15 p.m.	ASC S12, Noise

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3, and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
R. D. Hellweg, Jr., Chair P. D. Schomer, Vice Chair	**ISO/TC 43/SCI** Noise	ASC S12
R. W. Fischer, Chair	**ISO/TC 43/SC 3**, Underwater acoustics	ASC S1, ASC S3/SC 1, and ASCS12
W. Madigosky, Chair	**ISO/TC 108** Mechanical vibration, shock, and condition monitoring	ASC S2
M. L'vov, Chair	**ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures	ASC S2
D. D. Reynolds, Chair	**ISO/TC 108/SC4** Human exposure to mechanical vibration and shock vibration and shock as applied to machines, vehicles, and structures	ASC S2
D. J. Vendittis, Chair	**ISO/TC 108/SC5** Condition monitoring and diagnostics of machine systems	ASC S2
IEC		
C. Walber, U.S. Technical Advisor	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R. J. Peppin, Chair ASC S1
5012 Macon Rd., Rockville MD 20852

A. A. Scharine, Vice Chair ASC S1
U.S. Army Research Laboratory, Human Research & Engineering Directorate
ATTN: RDRL-HRG, Building 459 Mulberry Point Rd.
Aberdeen Proving Ground MD 21005-5425

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

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

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

Session 2aSPb

Signal Processing in Acoustics: Acoustic Detection, Localization, and Classification (Poster Session)

Kainam T. Wong, Chair
School of General Engineering, Beihang University, 37 Xueyuan Road, Haidian District, Beijing 100083, China

All posters will be on display, and all contributors will be at their posters from 10:45 a.m. to 11:45 a.m.

Contributed Papers

2aSPb1. MVDR beamforming using one bi-axial velocity-sensor. Yang Song (Nanyang Technolog. Univ., Singapore, Singapore) and Kainam T. Wong (School of General Eng., Beihang Univ., Xueyuan Rd., Beijing, China, ktwong@ieee.org)

A bivariate velocity-sensor measured two Cartesian components of the acoustic particle velocity field, which represents the spatial gradient of the acoustic pressure field. One such bi-axial unit provides azimuth-polar two-dimensional spatial directivity. This work presents the MVDR beam-pattern for such a sensing unit.

2aSPb2. The benefit of acoustic beamforming in solving the "cocktail party problem" for persons with acquired aphasia. Sarah Villard, Christine Mason, and Gerald Kidd (Speech, Lang., & Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, svillard@bu.edu)

Acoustic beamforming has been shown to improve identification of target speech in noisy listening environments for individuals with sensorineural hearing loss. This study examined whether beamforming would provide a similar benefit for individuals with aphasia (an acquired neurological language deficit). Persons with aphasia (PWA) are known to exhibit impaired language comprehension abilities; however, most work on language comprehension in PWA has been conducted in quiet settings, and little is known about the impact of competing auditory information on performance. In this study, we measured the intelligibility of target speech masked by other speech or noise for two presentation/microphone conditions: one condition, designated "KEMAR," provided natural spatial cues via KEMAR impulse responses; the second condition, designated "BEAM," enhanced the target speech level via a single-channel beamformer. In each condition, subjects heard a target sentence at 0 deg azimuth concurrent with two independent maskers—speech or speech-shaped speech-envelope-modulated noise—from ± 60 deg. Threshold target-to-masker ratios were measured adaptively for each subject, with individually-determined modifications in the procedures made for the subjects. Results indicated substantially lower (better) thresholds for the beamformer condition than for KEMAR, providing preliminary evidence that PWA may benefit from the use of acoustic beamforming in complex, multiple-source listening situations.

2aSPb3. Automatic environmental soundscape classification of continuous field recordings around Lake George, NY. Mallory M. Morgan (Rennselaer Polytechnic Inst., Greene Bldg, RPI, Troy, NY 12180, morgam11@rpi.edu), Vincent Moriarty (IBM, Bolton Landing, NY), and Jonas Braasch (Rennselaer Polytechnic Inst., Troy, NY)

The amount of audio data required for long-term bioacoustics monitoring is often too large to be manually sorted. Automatic environmental sound recognition techniques are therefore applied to extract relevant acoustic stimuli and classify these stimuli after a training period. In this work, a continuous 24-h, remote audiovisual recording system was developed and deployed near a tributary stream of Lake George, NY for the automatic collection of environmental and animal sounds. Besides monitoring natural environments, this system can also be used to establish an automatic protocol for collaborative business meetings. In conjunction with the efforts of the RPI/IBM Jefferson Project to create a network of sensors continuously monitoring the lake, the goal of this research is to automatically transcribe the soundscape of the lake using a network of directional microphone arrays positioned throughout the watershed. After establishing a training database of relevant acoustic stimuli, a convolutional neural network is used to classify unprocessed audio data. Pilot studies show good results when learning takes place directly from spectrograms, since the variability of non-speech acoustic events often requires more rich detail than can be provided by MFCCs or other compressed feature sets. [Work supported by NSF #1631674, CISL, and the Jefferson Project.]

2aSPb4. Bat biosonar echo analysis using spatial audio and wideband matched-filter techniques. Hyeon Lee (Mech. Eng., Virginia Tech, 100S Randolph Hall (MC0710), 460 Old Turner St., Blacksburg, VA 24061, hlee777@vt.edu), Chen Ming (Neurosci., Brown Univ., Providence, RI), Michael J. Roan (Mech. Eng., Virginia Tech, Blacksburg, VA), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Bats and dolphins show tremendous aptitude in hunting prey in difficult conditions. These conditions include large amounts of clutter, perhaps reverberation, and interfering signals/noisy backgrounds. Bats have been extensively studied to learn more about how they detect, track, and intercept prey. One important aspect of many of these studies is to understand the acoustics around the bat's head. In this work, a new reproduction methodology using spatial audio and matched-filter techniques will be presented. A custom-built tetrahedral 1st order soundfield microphone that captures high-frequency sound up to 80 kHz from all directions was developed to measure bat echoes in B-format. The three-dimensional echoes that the bat received, now in Ambisonic B-format were further processed using a wideband matched-filter. The matched-filter produces a wideband cross-ambiguity function (WAF) of received data and the transmitted signal for every 1 deg in azimuth and elevation. The thresholded matched-filter output provides precise estimates of bearing, elevation, range, and normal velocity. This poster will present details of the hardware and software development along with experimental results that illustrate the capabilities of this new approach.

Session 2pAA

Architectural Acoustics, Musical Acoustics, and Education in Acoustics: Higher Education Schools of Music

Brian Corry, Cochair
Kirkegaard Associates, 7733 Forsyth Boulevard, Suite 1100, St. Louis, MO 63105

Kirsten Hull, Cochair
Kirkegaard Associates, 801 W Adams St., Suite #800, Chicago, IL 60601

Chair's Introduction—1:30

Contributed Paper

1:35

2pAA1. The soundscape of music education. Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Lucky S. Tsaih (Architecture, National Taiwan Inst. of Sci. and Technol., Taipei, Taiwan), Hyun Paek, Marylin Roa, Jennifer R. Miller, Matthew Vetterick, and Keely Siebein (Siebein Assoc., Inc., Gainesville, FL)

Research on acoustical qualities of music education spaces based in soundscape theory provides a method to link the acoustical performance of a space with the instructional tasks that occur in the space with the architectural design features of the rooms. There has been limited study of the acoustical needs of teachers and students in music education spaces. Pirn (1978) showed that music practice rooms can become too loud as multiple musicians are placed in a closed and limited room volume without sufficient sound absorbing materials. Gade (1988) defined the characteristic of Support to measure cross room reflections from ceiling and wall surfaces to allow the musicians to hear each other. Tsaih (2011) found that the soundscape of music education was based on the ability of the students and teacher to be able to clearly understand spoken instruction and for students to hear each other and for the instructor to hear students playing in time, in tune and in dynamic. Case studies of music education spaces will illustrate how these concepts are integrated within the architectural design and acoustical analysis of the spaces.

Invited Papers

1:50

2pAA2. Lewis center for the arts at Princeton University. Joe Solway, Casey Eckersley, Joseph Digerness, and Raj Patel (None, 77 Water St., New York, NY 10005, joe.solway@arup.com)

The opening of Princeton's Lewis Center for the Arts in the Fall 2017 provided much needed performance and teaching spaces for the Theatre, Dance and Music programs at Princeton University, including an acoustically isolated orchestral rehearsal room, black box theater, dance theater, dance and acting studios, and music practice rooms individually hung from the roof slab above. Achieving the right balance of aural connectivity and sound isolation was an important part of the design, ensuring the Center feels vibrant and alive, while allowing teaching, rehearsal and performance to occur without disturbance. A key component to this work was the use of the Arup Soundlab, used to simulate the predicted acoustic environment within the music, drama and dance spaces, allowing the Princeton faculty and administration to listen to these spaces, and through discussion with Arup and architects from Steven Holl and BNIM, make informed decisions on the acoustic requirements for the design.

2:10

2pAA3. Glazer music performance center at Nazareth College. David Kahn (Acoust. Distinctions, 400 Main St., Ste. 600, Stamford, CT 06901, dkahn@ad-ny.com)

The new Glazer Music Performance Center at Nazareth College opened to rave reviews in Fall 2018. This project had several unique challenges including budget, schedule and a goal for simultaneous use of a warm-up room that sits within the footprint of the upper volume of the concert hall. The concert hall, while only seating 700, has a performance platform sized for a full symphony orchestra and choir. In order to acoustically accommodate these large ensembles, the room volume was set to 625 000 cubic feet. Since this would have required an unusually and impractically tall space, additional volume was captured above some of the hall's circulation areas, backstage, and above a few acoustically sensitive warm-up, teaching and practice spaces located behind the backstage area. Substantial sound isolation construction was developed to allow for simultaneous use. A second and related challenge was the project budget of $14M, translating to an average construction cost of $580/SF and included no low-cost spaces. A third challenge was an extremely aggressive project schedule requiring the facility to open in time for their 50th anniversary celebrations. All challenges were overcome thanks to an integrated design process carried out by talented and dedicated design and construction professionals.

2:30

2pAA4. DePaul University—Holtschneider performance center. Joseph W. Myers, Kirsten Hull (Kirkegaard Assoc., 801 W. Adams St. 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com), and Brian Corry (Kirkegaard Assoc., St. Louis, MO)

The DePaul School of Music has thrived for years in spite of teaching, rehearsal, and performance facilities previous facilities that were inadequate in terms of room acoustics, sound isolation and background noise. The new Holtschneider Performance Center in Chicago IL is the first and largest element in a three-phase design that utterly transforms the School of Music's facilities, finally giving the School facilities that match the quality of the faculty and students. The new building includes a 505 seat concert hall, three recital spaces, large rehearsal rooms, classrooms, numerous practice rooms and a recording suite. With such an extensive program the main challenge was to fit the full program on an urban site while respecting zoning limits to size and height. With careful organization, heavy concrete structure and thoughtful use of isolation joints, floating slabs and resilient walls and ceilings, the building provides excellent isolation despite some challenging adjacencies. The site drove an approach in which carefully controlled footprints for the major spaces are married to generous heights.

2:50

2pAA5. A multidisciplinary music and visual arts center at Earlham College. John T. Strong and Carl Giegold (Threshold Acoust., 141 W. Jackson Blvd., Ste. 2080, Chicago, IL 60604, jstrong@thresholdacoustics.com)

The Center for Visual and Performing Arts is a new construction on the campus of Earlham College, conceived as the new performance and rehearsal home for the Earlham Music department. It hosts a broad spectrum of ensemble types, including dedicated rehearsal spaces for jazz, percussion, and Javanese gamelan, and a full teaching studio and practice room suite. The flexible 260-seat Lingle Recital Hall hosts performances of small- to medium-sized ensembles; seating retracts to provide a flat floor rehearsal space for full orchestra and other large ensembles. An acoustically diffusive basket-weave wooden slat treatment around the lower perimeter of the room as well as variable acoustic systems spread throughout the hall allow a wide range of interior acoustic flexibility. The building also hosts a black box studio theater programmed by the Theater Arts department, and painting, textile, ceramics, and metalworking space for the Art department. Due to the building's compact footprint, and the high sound levels produced by the ensembles and art workspaces, extensive structural and architectural acoustic isolation measures are utilized. A centrally located recording suite with connectivity to the major performance and rehearsal spaces also functions as a teaching space for the recording arts.

3:10–3:25 Break

Contributed Paper

3:25

2pAA6. The evolution and creation of schools for music education. Hyun Paek, Gary W. Siebein, Marylin Roa, Jennifer R. Miller, and Matthew Vetterick (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, hpaek@siebeinacoustic.com)

The creation of a school of music can begin with a vision of one person or with the necessity felt by many. Case studies of two schools of music is presented which were conceived from very different approaches. Case study one shows an evolution of a complex building that started from the use of a low ceiling classroom designed for general instruction that is now a multistory facility that houses not only spaces for the entire music program at a college in Florida but also their entire fine arts program with a dormitory that encompasses the "live/learn" concept of the school. The second case study involves a first of its kind magnet school for the arts of a school district in Georgia that had no other facility to instruct talented students of music and fine arts on a vacant site. No instructors or directors of music was available for input in the programming and concept phases of the design and thus the design was left for the design team and the school district administrators to cooperate in generating the school's vision for the future.

Invited Papers

3:40

2pAA7. University of Colorado, Colorado Springs—Ent center for the arts. Matt Nichols (Jaffe Holden, 114A Washington St., Norwalk, CT 06854, mnichols@jaffeholden.com)

The UCCS Ent Center for the Arts is a transformational five-venue center; an innovative collaboration involving the university, six community arts partners and three local school districts. Jaffe Holden provided acoustic and audio/video systems design services for a wide range of spaces including a multi-use hall, recital hall, art gallery, drama theaters, art lab, recording studio, music rehearsal room, teaching studios and practice rooms. Our acoustic focus within the Shockley-Zalabak Theater was on making a successful multi-use hall to support a wide range of programming including full orchestra ensembles and small chamber groups. The acoustic response can be adjusted by a motorized system of drapes and acoustic banners concealed above a partially sound-transparent ceiling. A stage lift allows changes to be made to seating configuration and provides a stage extension. An acoustical shell can be configured to accommodate both large and small group performance. The Chapman Foundation Recital Hall's acoustics can also be tailored to suit various musical ensembles, thanks to an adjustable acoustic curtain system hidden above the partially sound-transparent ceiling and slatted wood wall at the rear of the stage. One of the biggest challenges was to successfully isolate a dance studio directly above a recording studio control room.

4:00

2pAA8. Armerding center for music and the arts at Wheaton College. Gregory A. Miller, Marcus R. Mayell, and Dawn Schuette (Threshold Acoust., LLC, 141 W. Jackson Blvd., Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

Wheaton College has long cultivated young musicians in a tight-knit community outside of Chicago. As the program has grown in recent decades, however, the community has burst at the seams in need of expanded space for faculty and students alike. The construction of a new science building on campus led to the availability of Armerding Hall to be redeveloped into a new home for the Conservatory on the central campus quad. Shallow floor-to-floor heights and limited structural capacities led to acoustic isolation strategies that provide "just enough" separation between faculty studios and relied on careful detailing for success. A steeply-raked Lecture Hall was reimagined to become a 100-seat recital hall with volume carved into adjacent spaces. Other spaces within the building include ensemble rehearsal rooms, a recording suite, and a room for multi-media composition. A second phase of the project will include new construction of a 650-seat concert hall and the college's first dedicated rehearsal hall for their formidable choral program.

4:20

2pAA9. Missouri State University—Ellis Hall. Brian Corry (Kirkegaard Assoc., 7733 Forsyth Blvd., Ste. 1100, St. Louis, Missouri 63105, bcorry@kirkegaard.com), Kirsten Hull, and Joseph W. Myers (Kirkegaard Assoc., Chicago, IL)

Missouri State University Ellis Hall is a 1957 modernist building originally designed for the music department. The building was mostly untouched over the years, despite poor sound isolation and unremarkable, undersized performance spaces. Kirkegaard worked closely with Patterhn Ives on a comprehensive renovation that included replacing the building mechanical systems and much of its exterior curtain wall. The interior is a virtual gut/rehabilitation, but wherever possible existing walls were incorporated into new sound-isolating assemblies to avoid unnecessary expense. The substantial scale of the project compared to the very limited budget required the design team to find carefully calibrated solutions for what to reuse and what to replace. One of the greatest challenges was to provide an acoustic superior recital hall, a goal that the School of Music emphasized as critical to the future of their program, from a renovation of the low-ceilinged existing recital hall. The solution was to demolish the hall's floor slab to capture the room volume from an under-used rehearsal room below, while preserving the upstage pipe organ and a handsome wood side wall. The result is a very successful new 240 seat recital hall in the shell of the old hall.

4:40

2pAA10. South Dakota State University. David Kahn (Acoust. Distinctions, 400 Main St., Ste. 600, Stamford, CT 06901, dkahn@adny.com)

Shortly after South Dakota set out to design their new performing arts center to support their music and theatre programs in 1997, they discovered the cost to construct these facilities was roughly double their budget. After 6 years of planning and design, the phase 1 performing arts center opened. Then, after an additional 5 years, the University was able to raise enough money to construct the second phase of their performing arts center which included a state—of-the-art large proscenium theatre, rehearsal rooms for choir and large instrumental ensembles, a jewel box recital hall with a pipe organ, music teaching studios and practice rooms. The accommodation of the pipe organ into the design of the recital hall was especially challenging because the pipe organ was originally designed and installed in a much larger and different-shaped church. Pipe organ music was not a high priority for the use of the new recital hall, and therefore some special accommodations were made to improve the sound of the pipe organ without compromising the acoustics for the room's primary use as a recital hall.

Session 2pAB

Animal Bioacoustics and Signal Processing in Acoustics: Bioinspiration and Biomimetics in Acoustics II

Rolf Müller, Chair
Mechanical Engineering, Virginia Tech, ICTAS Life Sciences District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061

Chair's Introduction—1:00

Invited Paper

1:05

2pAB1. The evolution of bat robots. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu), Roman B. Kuc (Elec. Eng., Yale, New Haven, CT), and Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Tech, Roanoke, VA)

The extraordinary skills of bats in supporting dexterous mobility in complex environments based on just two pulsed trains of one-dimensional biosonar echoes has attracted attention from engineers for several decades already. To explore whether it is possible to reproduce at least certain of these capabilities, a diverse set of "bat robot" prototypes have been built. The earliest, most basic of these systems were limited to estimating the distance of sonar targets based on the acoustic time-of-flight. From there, systems improved to take advantage of more echo waveform features, e.g., for target recognition. Two (or more) receivers were introduced to exploit binaural differences, e.g., for target tracking. Rigid ear rotations served to enhance the signal-to-noise ratio be focusing on a target or to determine target direction from the echo amplitudes received across a scan. Biomimetic emission and reception baffle shapes, i.e., "noseleaves" and "pinna," were added to narrow the sonar beams and create direction-depended spectral signatures. Deformations of flexible baffle structures that mimic the muscular actuation of the noseleaf and pinna shapes seen in bats have been added to these systems. Mobility of the entire systems has been provided by mounting them on pan-tilt units, robot arms, mobile robots, and drones.

Contributed Papers

1:25

2pAB2. A soft robotic actuation system for a bat robot. Emmet Eckman, Matthew Conk, Sierra Lunsford, Charlie Martin, Agoshpreet Singh, Kent Sullivan, Yanhao Wang (Mech. Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Circle, Blacksburg, VA 24060, eckman19@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

A crucial development in the field of biomimicry is accurately recreating the kinematics of various organisms. In the example of a bat robot in development inspired by the greater horseshoe bat, *Rhinolophus ferrumequinum*, this takes the form of accurately recreating fast motions of the pinnae and noseleaf observed during echolocation. In order to accurately recreate these baffle shapes it is important to choose a material that is malleable but will retain its originally molded state. An actuation system was developed to replicate the ear movement seen in the bat when using biosonar. Several actuation designs were tested, including a physical motor connection, shape memory alloys, and soft-robotic pneumatics. In addition to the actuation system, a feedback system was developed in order to accurately control the robot and provide information which could be used to determine the effectiveness of the robot. Additionally, the final challenge to assemble all the components, i.e., the baffle shapes, actuator mechanism, and feedback control mechanism, in a way that recreates the baffle motions in an effective and accurate manner.

1:40

2pAB3. Design and manufacturing of a miniature pneumatic actuator for a bat robot. Kent Sullivan, Matthew Conk, Emmet Eckman, Sierra Lunsford, Charlie Martin, Agoshpreet Singh, Yanhao Wang (Mech. Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Circle, Blacksburg, VA 24060, kntsully@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

There have been many pneumatic actuator designs created over the past several years. One such design consists of multiple, completely separated air cells that when pressurized, expand and push off one another causing the actuator to bend. With this chamber design, the actuator requires a small change in volume to deform, effectively leading to improved reliability and increased actuation speed. Other design characteristics such as cross section geometry, inner/outer wall thickness ratio, distance between cells, cell height, and material selection can be modified to optimize the actuator's performance. Incorporating this design into the bat robot allowed for greater soft robotic ear deformation, however it is too large for this application. To see if it was possible to reduce the size of the actuator while maintaining its functionality, several adjustments were iteratively made to previously described mold designs. Manufacturing remained a two-step process, but removable side walls and a filter-ventilation system were added to allow for the silicone, used to cast the actuator, to fully cure. Ultimately, the actuator produced maintained its expected functionality and was produced at a smaller scale than originally thought possible.

1:55

2pAB4. Acoustic emission system for a bat robot. Nathan Cox (Aerosp. and Ocean Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Circle, Blacksburg, VA 24060, nathc17@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

One of the more difficult challenges in producing a biomimetic bat robot is replicating the bats' biosonar emission capabilities. For mimicking horseshoe bats (Rhinolophidae), this means creating a high-power acoustic source that can also properly illuminate the noseleaf structure over a bandwidth of approximately 20 kHz. Computer tomography (CT) scans were used to inform an accurate noseleaf model which was then abstracted from life using three-dimensional mesh software. A waveguide was designed to direct sound from two electrostatic transducers into a small nozzle, creating a point source. Numerical acoustic simulations were used to optimize these waveguide shapes to obtain a good compromise between point source behavior, high output amplitudes, and minimal internal reverberation. Physical geometries were procedurally generated to create a three-dimensional design space with parameters eccentricity, length and convexity. Data from each waveguide were analyzed for volume, multidirectionality, and echoing. A second round of simulations was executed to determine the effect of nozzle width on the output. Findings indicated that waveguide convexity and eccentricity played primarily into echoing and increased length reduced output volume. As nozzle diameter increased, the output volume increased, but sound became unidirectional. Future research may include development of alternative emission systems without a need for waveguides.

2:10

2pAB5. Bat-board: An integrated electronic system for a robotic bat. Lucas Mun (Elec. and Comput. Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Cir, Blacksburg, VA 24061, cmun09@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

The biosonar of horseshoe bats combines the standard elements of any sonar system, i.e., pulse generation and echo processing, with a peripheral dynamics provided by deforming baffle shapes for ultrasonic emission ("noseleaves") and reception (ears). Realizing a "bat robot" to reproduce this biological system requires the implementation of three main electronic functionalities: power distribution, analog signal conditioning, and robot control. To meet the size constraints of the bat robot, the work presented here has designed/implemented a "Bat Board" that integrates all three of these functionalities within a single circuit board. The power distribution component of the Bat Board supplies the high voltage amplifier, the air compressors/valves for the pneumatic actuators in the noseleaf/the ears of the sonar head, and the data acquisition systems. The high voltage signal amplification component was designed to power the electrostatic transducers that were selected for their broad-band output. To combat interferences originating from the power distribution and the digital control signals, different PCB layout paradigms were implemented in the design process such as separating ground planes and the extensive use of bypass capacitors. The result of this efforts has been a single PCB implementation with a very compact footprint and the potential for further miniaturization.

2:25

2pAB6. Data acquisition and controls for a bat robot. Brandon Walker (Elec. and Comput. Eng., Virginia Polytechnic Inst. and State Univ., 1075 Life Sci. Cir, Virginia Tech, Blacksburg, VA 24061, branw@vt.edu), Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., Roanoke, VA), and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

Robotic reproductions of the dynamic and adaptive nature of the biosonar systems found in many bat species require a capable back-end that is capable of integrating mechanical, acoustical, electrical, and computational functions in an efficient manner. This is particularly true when mimicking bat species that change the shapes of their noseleaves and pinna as part of their biosonar behaviors. To address the challenge of replicating these highly integrated functions, a platform consisting of commercial, off-the-shelf components, centered around a microcontroller (Arduino Due) and a single-board computer (Raspberry Pi 3) has been designed. The real-time microcontroller has been delegated the signal generation for the ultrasonic pulses and data acquisition for the echoes, as well as all control operations for the mechanical periphery of the robotic bat head. This includes synthesizing output waveforms, conditioning measured data, and maintaining the state of the pneumatic actuation systems. Commands are issued directly by the computer, which is responsible for orchestrating overall "behaviors" and managing the relevant data. In addition, the control system acquires and stores meta data for the echoes such as geospatial location and acquisition time. Future improvements to this system will seek to establish closed loop control liking echo analysis to peripheral dynamics.

2:40

2pAB7. Integrating a brain into a bat robot. Mohammad Omar Khyam (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, mok@vt.edu), David Alexandre (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA), Ananya Bhardwaj, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

The sophisticated biosonar systems of horseshoe bats have enabled these animals to navigate and pursue prey in complex environments. A conspicuous peripheral dynamics in which the animals' noseleaves and pinnae change during biosonar behaviors could play an important role in enabling these capabilities. It may be hypothesized that for the integration between peripheral dynamics and neural signal processing/estimation to be maximally effective, the periphery should be controlled by feedback from the output of the subsequent neural echo processing. In the way, the specifics of sensory information encoding in the periphery could be controlled based on the needs of the neural signal processing. As a first step towards such an integration in a biomimetic sonar head, a computational model for the inner ear and the auditory nerve's spike code has been integrated with a dynamic periphery that—like the computational models—mimics horseshoe bats. For each model stage, alternative versions with different levels of complexity have been implemented to test how module complexity and the values of the associated parameters affect the capacity of the echo representation to encode sensory information. These effects have been tested based on a large dataset of 220 000 echoes collected in natural forest environments.

2:55

2pAB8. Sonar signal representation mimicking the inner ear of horseshoe bats. Mohammad Omar Khyam, Ananya Bhardwaj (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, mok@vt.edu), David Alexandre (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Current models of the initial stages of auditory processing in mammals usually agree that the input signals are split into a bank of bandpass filters. However, the available models differ substantially in their level of complexity and the number of parameters needed. In the current work, the varying levels of complexity in filterbank models of the basilar membrane have been evaluated in the context of modeling the cochlear processing of natural biosonar echoes in horseshoe bats. To this end, three different types of filterbank models have been implemented to represent the range of complexity spanned by models that are in use for the human inner ear: The simplest model, a gammatone filterbank, is a linear model with symmetric filter transfer functions. The gammachirp filterbank is also linear, but mimics the asymmetric transfer functions of the basilar membrane. Finally, the dual resonance nonlinear (DRNL) model adds a level-dependent behavior. Here, all three models have been adapted to the specifics of the basilar membrane of horseshoe bats which is characterized by an "auditory fovea" with exceptionally high filter qualities. The outputs of the different models have been encoded into a sequence of neural spike times before being evaluated with various information-theoretic methods.

3:10

2pAB9. Biomimetic spike representation for the encoding of sonar signals. David Alexandre (Aerosp. Eng., Virginia Tech, ICTAS II, 1075 Life Sci. Circle, Blacksburg, VA 24060, davida49@vt.edu), Mohammad Omar Khyam, Ananya Bhardwaj, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Biosonar echoes received by bats in their natural habitats are short, highly time-variant acoustic waveforms. Because of these signals properties, the way in which the bats' auditory nerve represents the echoes could be a model for how sparse neural time-code can capture salient signal features. Here, the spike-encoding of natural echoes has been studied based on a large data set containing about 220 00 echoes that were collected by hand-carrying a biomimetic robotic sonar head through forest environments. The sonar head was equipped with flexible noseleaf and pinna shapes that could deform during pulse emission/echo reception in a similar fashion to what horseshoe bats do. This peripheral dynamics was turned on for half of the recoded echoes and turned off for the other half. The echo waveforms were transformed into spike trains using two spike-generation models, each with different levels of complexity: The simplest version was a "leaky integrate-and-fire model; in the more complex version, a response kernels was added to model the refractory behavior of the neurons. The input to the spike generation models were the outputs of three different basiliar membrane models of varying complexity levels. The coding capacity of the spike trains has been evaluated using information-theoretic methods.

3:25

2pAB10. Information-theoretic evaluation of brain-inspired sonar signal representation. Ananya Bhardwaj, Mohammad Omar Khyam (Mech. Eng., Virginia Tech, ICTAS II, 1075 Life Sci. Circle, Mail Code 0917, Blacksburg, VA 24061, ananya22@vt.edu), David Alexandre (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Many bat species, such as horseshoe bats, move their external auditory periphery (noseleaf and pinnae) during emission/reception of their biosonar signals. This peripheral dynamics has already been shown to enhance sensory information encoding at the level of the analog echo waveforms. However, the bats' brains do not process any analog waveforms directly but rely on spike codes generated in the auditory nerve instead. Hence, it is desirable to evaluate the effect of the peripheral dynamics at the level of these spike codes. As input for this analysis, natural foliage echoes akin to the forest environments experienced by bats were recorded using a biomimetic robot with a dynamic periphery similar to that of horseshoe bats. The echoes were converted into neural spike trains through a signal-processing model of the bats' cochlea/auditory nerve. The effect of the dynamic periphery was then investigated using information-theoretic techniques. As a first step, entropy was estimated to quantify coding capacity. The results showed an increased entropy within signals corresponding to a dynamic periphery when compared to a static periphery. For the next steps, the variability in characteristic spike train features such as spike timing, spike rates, and spike intervals will be characterized with additional information-theoretic methods.

3:40–3:55 Break

3:55

2pAB11. Bat biosonar study using spatial audio techniques. Hyeon Lee (Mech. Eng., Virginia Tech, 100S Randolph Hall (MC0710), 460 Old Turner St., Blacksburg, VA 24061, hlee777@vt.edu), Chen Ming (Neurosci., Brown Univ., Providence, RI), Michael J. Roan (Mech. Eng., Virginia Tech, Blacksburg, VA), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Bats have highly specialized hearing adaptations that give them target detection, localization, and interception abilities that far surpass any man-made SONAR. There is a large body of research that has studied and attempted to mimic this remarkable natural sonar system using various methodologies. In many studies, multiple microphones and time difference of arrival were used for bat ultrasonic wave measurements. In this work, a new method using spatial audio techniques is introduced for bat biosonar measurements and analyses. A custom-built tetrahedral soundfield microphone that captures high-frequency sound up to 80 kHz from all directions was developed to acquire bat ultrasonic echoes in Ambisonic B-format. The recorded sound was processed using the HARPEx (High Angular Resolution Planewave Expansion) technique that decodes the 1st order spherical harmonics of the soundfield at the microphone into two plane waves in each time-frequency bin. The use of the soundfield microphone and associated processing provides new tools for analysis of bat echoes from the perspective of the bat. This talk will present details of the hardware and software development along with experimental validation results.

Invited Paper

4:10

2pAB12. Bio-inspired generalizable-focusing wideband FM sonar. James A. Simmons and Chen Ming (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu)

The bat's inner ear and auditory brainstem register successive frequencies in FM sweeps of broadcasts and echoes with single spikes. Using the neural spectrogram of each broadcast as reference, spike latencies trace the FM sweeps for imaging echo delay by spectrogram correlation. If mutually-interfering glint reflections are present, the echo's neural spectrogram has regularly-spaced ripples caused by amplitude-latency trading. Glint-delay estimates are generated by transforming the ripple pattern from frequency to time in real time using feedforward inhibition. Several glint delay estimates can coexist in focused target images before accumulation of nulls generates too many glint estimates and blurs the image. Echo lowpass filtering, characteristic of clutter, affects a wide swath of frequencies and changes the slope of the neural spectrogram, which evokes multiple nulls and causes image blurring that suppresses the clutter. Echo Doppler shifts also change the slope of the FM sweep, and the same mechanism is capable of blurring Doppler-shifted images so that troublesome ambiguity is suppressed. By internally adjusting the neural FM sweep of the broadcast reference, the images of echoes can be refocused onto any desired location not only in range, azimuth, and elevation, but also on the range-Doppler plane. [Work supported by ONR.]

4:30

2pAB13. Accommodating constraints for modeling of biosonar processing by big brown bats. Chen Ming, James A. Simmons (Dept. of Neurosci., Brown Univ., 185 Meeting St., Providence, RI 02912, chen_ming@brown.edu.edu), Stephanie Haro (Program in Speech and Hearing Sci. and Technol., Harvard Med. School, Boston, MA), and Jason E. Gaudette (Sensors and Sonar Systems Dept., Naval Undersea Warfare Ctr., Newport, RI)

Constraints on modeling of wideband FM biosonar are acoustic (small targets return discrete replicas of the incident sound from individual component parts), auditory (inner-ear transduction segments broadcasts and echoes into numerous, parallel bandpass channels with integration-times of ~350 μs), neural (processing retains parallel frequency bands throughout the auditory pathway that use single spikes to mark frequencies), and perceptual (bats nevertheless perceive true delays of individual glint reflections, not just spectral coloration). The bat's target model is geometric, registering the glints binaurally along the range axis. However, because closely-spaced glints are represented by their mutual interference, not by separate acoustic reflections, the ripples have to be transposed from an interference pattern back into glint delay estimates. Auditory cortical neurons selective for specific patterns of ripple come to register instead the underlying time separation of the glint reflections. The developing SCAT model of wideband FM biosonar incorporates a network for glint delay transformation that also offers a route for clutter suppression by contrasting dispersed, unfocused clutter images with tightly focused target images in range and azimuth space. The feedforward inhibition of this network preserves real-time operation for target reconstruction as well as for guidance in clutter. [Work supported by ONR.]

4:45

2pAB14. Biomimetic foliage echo simulation. Michael Goldsworthy (Comput. Sci., Virginia Tech, 155 Otey St., Rm. 323, Blacksburg, VA 24061, michaeljg@vt.edu) and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Many echolocating bat species are capable of navigating through highly cluttered environments, such as dense foliage—apparently without difficulty. Since interpreting foliage echoes must hence be key to the animals' navigation capabilities, simulating such echoes would be a major step towards understanding bat navigation. This is a difficult task, as vegetation echoes are highly complex and stochastic signals. Prior work on modeling foliage echoes frequently relied on physical approximations, since full physics models are computationally infeasible. For simulating the echo from a leaf, a round disk approximation has been used, since it's echo has a known physical solution. These disks have been distributed according to the locations of leaves in a tree using a Lindenmayer System (L-System), thus creating a full tree echo model. Successful prior work in using a machine learning approach for echo based tree classification, shows that a data driven approach may also be useful in modeling echoes. Hence, future research will be using machine learning methods, such as auto-encoders and generative adversarial networks, to simulate the echoes of leaves and trees. The echo data to be fed into these methods will be of leaves recorded in an anechoic chamber using a biomimetic sonar head.

5:00

2pAB15. Biomimetic solutions to finding passageways in foliage. Ruihao Wang and Rolf Müller (Mech. Eng., Virginia Tech, 112 Hearthstone Dr., Apt. 210, Blacksburg, VA 24060, ruihaow1@vt.edu)

Many bats species live in densely vegetated habitats. Hence, they must have evolved the ability to detect passageways through the foliage. To identify echo features that bats have at their disposal to accomplish this sensing task, a biomimetic sonar head was used to ensonify artificial foliages in the laboratory. These foliages consisted of plastic vines that were arranged to create gaps of defined width and height. The conventional sonar approach to detecting gaps between scatterers is based on the drop in energy that occurs when the sonar beam is aimed at a gap. However, the performance of such an energy-detector decrease as the sonar-beam width increases relative to the angle subtended by the gap edges. Hence, reliable detection of a narrow gap with a wide beam and/or from a distance is not feasible with this approach. Here, a machine-learning approach based on convolutional neural networks was employed to identify features in energy-normalized spectrogram that could support passageway-detection independently of echo energy. The results indicate that features other than echo energy exist in the spectrogram that by themselves support much better passageway detection performance than the energy-based reference. Work to understand the features learned by the convolutional neural network is currently underway.

5:15

2pAB16. Biomimetic sonar and the problem of finding deterministic targets in foliage. Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Polytechnic Inst. and State Univ., 3719 Parliament Rd., Apt. 22, Roanoke, VA 24014, josephs7@vt.edu) and Rolf Müller (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

The ability of (bio)sonars to find targets-of-interest is often hampered by a cluttered environment. For example, naval sonars encounter difficulties finding mines partially or fully buried within sand. Such situations pose target identification challenges are much harder than target detection and resolution problems. There are many bat species which navigate and hunt in dense vegetation and thus must be able to identify targets-of-interest within clutter. Evolutionary adaptation of the bat biosonar system is likely to have resulted in the "discovery" of features that support making distinctions between clutter and echoes of interest. The most well-established case is given by cf-fm bats that use Doppler shifts caused by the wingbeat of a flying insect prey to identify the prey in foliage. Other bat species have been shown to use a passive sonar approach that is based on unique prey-generated acoustic signals. Some of the most interesting cases can be found in bat species that are successful in finding preys that apparently does not emit any distinguishing sounds themselves and would hence limit the bats to an active-sonar approach. Such bat species could provide model species for new ways in which the target-identification problem in clutter can be solved with active sonar.

Session 2pBA

Biomedical Acoustics and Signal Processing in Acoustics: Cardiovascular Ultrasound: Imaging and Therapy II

Kevin J. Haworth, Cochair
University of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

Jonathan A. Kopechek, Cochair
University of Louisville, 2301 S Third St, Lutz Hall, Room 400, Louisville, KY 40292

Invited Papers

1:00

2pBA1. Ultrasound-targeted microbubble destruction (UTMD)—Novel applications for microRNA delivery. Howard Leong-Poi (Cardiology/Medicine, St. Michael's Hospital, 6-044 Queen Wing, 30 Bond St., Toronto, ON M5B1W8, Canada, leong-poih@smh.ca)

Ultrasound-targeted microbubble destruction (UTMD) is a non-invasive technique for gene delivery, utilizing high power ultrasound and nucleic acid-bearing microbubbles. UTMD has been used in a variety of *in vivo* applications, including cardiac and skeletal muscle, kidney, liver, cerebral and even lung, and have been studied using many gene vectors, including plasmid, viral and small interfering RNA. The focus of gene therapy has now shifted towards small non-coding RNAs, including microRNAs (miRNA). These non-coding RNAs are important transcriptional and post-transcriptional inhibitors of gene expression that regulate the translational output of target messenger RNAs. MicroRNAs have been shown to participate in a multitude of cellular process, and their dysregulation may play an important pathophysiologic role in many different human pathologies. Coupled with their ability to specifically target particular cellular pathways, makes the possibility of exploiting miRNAs to develop therapeutic strategies extremely attractive. This presentation will focus specifically on several miRNAs with UTMD applications in (1) chronic ischemic peripheral arterial disease, (2) ischemia-reperfusion injury, and (3) abdominal aortic aneurysms (AAA). For each application, we will discuss selection of miRNA, aspects of tissue targeting *in vivo*, enhancement of therapeutic effect and potential for clinical translation.

1:20

2pBA2. Lipid-shelled microbubbles for ultrasound-triggered release of bioactive gases to treat stroke and cardiovascular disease. Christy K. Holland, Himanshu Shekhar, and Maxime Lafond (Internal Medicine, Div. of Cardiovascular Diseases and Biomedical Eng., Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Ischemia-reperfusion-induced neurological injury is a primary cause of stroke disability. Xenon (Xe), a bioactive gas, has potential as an effective and nontoxic neuroprotectant for the treatment of ischemic stroke. Nitric oxide (NO) is a potent bioactive gas capable of inducing vasodilatory, anti-inflammatory, neuroprotectant and bactericidal effects. The goal of this work was to develop lipid-shelled microbubbles for site-specific release of Xe or NO upon pulsed ultrasound exposure. Gas-loaded microbubbles were synthesized by high-shear mixing of a lipid dispersion in a vial that contained, Xe or NO, and octafluoropropane (OFP) in combination. Attenuation spectroscopy measurements demonstrate the feasibility of 6-MHz pulsed Doppler ultrasound-triggered release of Xe or NO from microbubbles. The addition of OFP in the lipid-shelled microbubbles increased the number density, size, and stability of the microbubbles, particularly in undersaturated saline. Gas chromatography mass spectrometry was employed to measure Xe dose (127 ± 29 μl Xe/mg lipid). The payload of NO in the microbubbles (97 ± 12 μl NO/mg lipid) was assessed using an amperometric sensor. Intravenous administration of microbubbles carrying a neuroprotective or a vasodilatory gas in combination with ultrasound exposure has potential as a novel noninvasive strategy for local therapeutic delivery to modulate the effects or duration of cerebral ischemia.

Contributed Papers

1:40

2pBA3. Remote implantation of ultrasound responsive multi-cavity microparticles for early atherosclerosis. Xiaoqian Su, Reju G. Thomas, Lakshmi Deepika Bharatula, and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technol. Univ., 62 Nanyang Dr., Block N1.2, 01-06, Singapore 637459, Singapore, jameskwan@ntu.edu.sg)

Atherosclerosis is an inflammatory disease of arteries, and results in stroke or heart attacks—the leading causes of death and disabilities in the developed world. Yet drug treatments for this chronic inflammatory disease remains to be addressed. Here, we developed a multi-cavity poly-co-lactic-co-glycolic microparticles (mc-PLGA MPs) capable of being remotely implanted at the site of arterial injury with focused ultrasound. The obtained mc-PLGA MPs present with two to five submicron surface cavities with rough inner surfaces. After exposure to high intensity focused ultrasound (HIFU) in an agarose tissue phantom, mc-PLGA MPs extravasated beyond the lumen of the channel at an average distance of 4.29 ± 1.19 mm, and sustained release of rhodamine-B for up to 15 days. Similarly, mc-PLGA MPs

were able to be implanted into the sub-endothelial space of an *ex vivo* porcine artery model without observable damage to the artery. To further validate the drug release, we exposed a foam cell spheroids model to mcPLGA MPs and HIFU, and show that particles were dispersed throughout the spheroids and sustained delivery throughout. The results here highlight the potential for HIFU-guided implantation of mcPLGA MPs to improve local and sustained treatment of inflamed arterial tissue.

1:55

2pBA4. Detection of nucleic acid-loaded microbubbles in mouse hearts during ultrasound-mediated delivery. Meghan R. Campbell, Mariah C. Priddy, and Jonathan A. Kopechek (BioEng., Univ. of Louisville, 2301 S Third St., Lutz Rm. 419, Louisville, KY 40292, meghan.campbell@louisville.edu)

Heart disease is a leading cause of death worldwide. Recently, there has been a growing interest in nucleic acid-based therapeutics, such as microRNAs (miRs) or microRNA inhibitors (antimiRs), for cardiac repair. Previous studies have shown that regulating microRNA levels in the heart can promote proliferation of cardiac cells, decrease fibrosis, and improve cardiac function. Ultrasound targeted microbubble destruction (UTMD) is in development to focus the delivery of therapeutic nucleic acids to the heart and reduce adverse systemic effects. Microbubbles can be loaded with nucleic acids and exposed to ultrasound in order to induce cavitation and enhance delivery at the target site. In this study phospholipid-coated microbubbles were loaded with therapeutic miR mimics or antimiRs and injected intravenously in mice. Ultrasound pulses (2.5 MHz, 0.9 MPa peak negative pressure) were applied to the heart using a P4-1 array on a Verasonics Vantage ultrasound system. The ultrasound images were analyzed to detect microbubbles in the heart during treatment. Increased mean intensity during infusion was associated with increased delivery of nucleic acids to the heart as assessed with qPCR. The results suggest that quantitative analysis of ultrasound images to detect microbubbles *in vivo* may aid in monitoring UTMD treatment for improved cardiac health.

2:10

2pBA5. Pentagalloyl glucose effects on murine abdominal aortic aneurysms. Jennifer L. Anderson, Alycia G. Berman, Elizabeth E. Niedert (Biomedical Eng., Purdue Univ., 206 S Martin Jischke Dr., Rm. 3083, West Lafayette, IN 47907, ander934@purdue.edu), Sourav Patnaik, Ender A. Finol (Mech. Eng., The Univ. of Texas at San Antonio, San Antonio, TX), and Craig J. Goergen (Biomedical Eng., Purdue Univ., West Lafayette, IN)

An abdominal aortic aneurysm (AAA) is a dilation of the abdominal segment of the largest artery in the body and can be accompanied by significant risk of rupture and mortality. The current treatment, surgical repair, carries risks and complications. As such, there is need for less invasive therapies capable of curbing aneurysm growth. Here we use an AAA mouse model to evaluate the potential of pentagalloyl glucose (PGG) to suppress aneurysm growth. To induce aneurysms, 5.0 mg/mL pancreatic porcine elastase (PPE) was topically applied to the infrarenal aorta and 0.2% beta aminoproprionitrile was continuously administered via drinking water. Four of the eight animals also received topical treatment of 0.06% PGG via gauze before PPE treatment. High frequency ultrasound imaging (Vevo2100 system, VisualSonics) with a MS550D transducer (40 MHz center frequency) was performed prior to surgery, and every week thereafter for 4 weeks. PGG-treated animals had a small but nonsignificant decrease ($p = 0.09$) in effective maximum aortic diameter on days 7 and 14 that was not present on day 28. Current work is being performed to quantify PGG binding via histological characterization. Ultimately, we aim to investigate the parameters required for PGG to be an effective treatment for mechanically-stabilizing small AAAs.

2:25–2:40 Break

Invited Papers

2:40

2pBA6. Measuring thrombolysis in a static model to assess ultrasound protocol efficacy. Curtis Genstler and Misty L. Noble-Vranish (BTG plc / EKOS Corp., 11911 N. Creek Pkwy S, Bothell, WA 98011, curtis.genstler@btgplc.com)

Over the years, EKOS has developed an efficacy measure called Lysis Enhancement Factor (LEF) to evaluate the performance of prototype ultrasound catheters and acoustic protocols. LEF is defined as the percent increase in the rate of lysis compared to tPA alone as measured in our simple clot model consisting of a plasma clot in a static tube. The amount of thrombolysis can be measured by determining the change in fibrin weight compared to control clots. The LEF is expressed as a percent change in the thrombolysis rate compared to lytic control. When the EkoSonic catheter was initially launched, the *in vitro* LEF was about 50% using a constant amplitude pulsed ultrasound protocol with an average power of 2.7 W per transducer group. Later, the MACH4 ultrasound protocol was introduced with an LEF of about 70%. This protocol had a higher average power (3.5W) and consisted of variable pulse amplitudes, which were randomly sequenced. Further investigations with the clot model lead to the development of the currently marketed MACH4e protocol with an LEF of about 90% but with variable amplitude pulses, which cycle up and down. The clot model and LEF are being used to further develop EKOS' technologies.

3:00

2pBA7. Histotripsy-enhanced thrombolysis. Kenneth B. Bader (Radiology, Univ. of Chicago, 231 Albert Sabin Way, CVC 3935, Cincinnati, Ohio 45267-0586, Kenneth.Bader@uc.edu), Samuel A. Hendley (Graduate Program in Medical Phys., Univ. of Chicago, Chicago, IL), Viktor Bollen (Radiology, Univ. of Chicago, Chicago, IL), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Kevin J. Haworth, and Christy K. Holland (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Deep vein thrombosis (DVT) is a major public health problem, affecting 600,000 Americans annually with a healthcare cost of $10 billion. Standard interventional techniques are not effective for the chronic thrombus components present in 27% to 43% of DVT cases. Histotripsy is a focused ultrasound therapy that employs the mechanical action of bubble clouds to ablate tissue and induce vigorous fluid mixing. We have demonstrated synergy between histotripsy and the thrombolytic recombinant tissue plasminogen activator (rt-PA) for dissolution of retracted venous clots *in vitro*. Here, the role of histotripsy-induced bubble activity in rt-PA clot dissolution will be

discussed. The rt-PA thrombolytic efficacy of histotripsy pulses that nucleate bubble activity either via (1) shock wave scattering or (2) exceeding the intrinsic cavitation threshold of the clot will be presented. The impact of each histotripsy type on the formed elements that comprise the clot will be described, as well as the likelihood of containing mechanical activity within the target zone. Finally, the utility of passive cavitation imaging to quantify the bubble activity necessary for liquefaction of tissue-mimicking phantoms representative of acute and chronic pathologies will be presented.

Contributed Papers

3:20

2pBA8. Design of a focused ultrasound transducer for histotripsy-thrombolytic combination therapy. Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, 1013 NE 40th St., Seattle, WA 98105, amax38@u.washington.edu), Kevin J. Haworth, Christy K. Holland (Div. of Cardiovascular Health and Disease, Univ. of Cincinnati, Cincinnati, OH), and Kenneth B. Bader (Dept. of Radiology, Univ. of Chicago, Chicago, IL)

Chronic thrombus components are resistant to removal by current interventional techniques and can act as thrombogenic sources. Histotripsy is a focused ultrasound therapy that utilizes the mechanical activity of bubble clouds to liquefy target tissues. *In vitro* experiments have demonstrated histotripsy provides enhancement of the thrombolytic agent recombinant tissue plasminogen activator in a retracted clot model representative of chronic thrombus. Although these *in vitro* results are promising, further refinement of the acoustic source is necessary for *in vivo* studies and clinical translation. To define the source parameters for use *in vivo*, a design study was conducted with transcutaneous exposure of porcine and human iliofemoral deep venous thrombosis (DVT) as the target. Design parameters were selected to confine the focus, and thus bubble activity, within the femoral vein of DVT patients. Furthermore, the therapy array accommodates the placement of a confocal diagnostic linear array for image guidance. Based on the design criteria, a 1.5-MHz elliptical source with 6-cm focal length and a focal gain of 70 was selected. Details of the design, fabrication, and characterization the source will be presented, as well as the means by which bubble activity is initiated.

3:35

2pBA9. Intrinsic threshold versus shock scattering histotripsy treatment of clots *in vitro*. Viktor Bollen (Dept. of Radiology, Univ. of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301A, Chicago, IL 60637, bollen@uchicago.edu), Kevin J. Haworth (Internal Medicine, Div. of Cardiovascular Diseases and Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Christy K. Holland (Internal Medicine, Div. of Cardiovascular Diseases and Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), and Kenneth B. Bader (Radiology and Committee on Medical Phys., Univ. of Chicago, Chicago, IL)

Catheter-directed thrombolytics are the frontline therapy for iliofemoral deep vein thrombosis. We have demonstrated that histotripsy can enhance the efficacy of the thrombolytic recombinant tissue plasminogen activator (rt-PA) through the mechanical activity of bubble clouds. Depending on the histotripsy pulse duration, bubble activity can be initiated via shock wave scattering or exceeding the peak negative pressure of the intrinsic cavitation threshold. In this study, the thrombolytic efficacy and bubble activity of shock scattering and intrinsic threshold histotripsy were compared. The mass loss was assessed of retracted human whole blood clots exposed to rt-PA (0 or 2.68 μg/mL) and histotripsy pulses of 1 (intrinsic threshold) or 5 (shock scattering) cycles duration. During the histotripsy exposure, bubble cloud emissions were captured on a linear array and processed to form passive acoustic images. The combination of rt-PA and histotripsy was more efficacious than rt-PA alone at 25 MPa peak negative pressure for single-cycle pulses, and 20 MPa peak negative pressure for five-cycle pulses. The degree of thrombolytic efficacy correlated with bubble cloud emissions within the clot. Overall, these results indicate that rt-PA and shock scattering or intrinsic threshold histotripsy is a promising combination thrombolytic therapy.

3:50

2pBA10. *In vitro* assessment of the relationship between medium stiffness and bubble activity for histotripsy-induced liquefaction. Samuel A. Hendley, Gregory Anthony (Committee on Medical Phys., Univ. of Chicago, 5812 S Ellis Ave., IB-016, Chicago, IL 60637, hendley@uchicago.edu), Viktor Bollen, and Kenneth B. Bader (Radiology, Univ. of Chicago, Chicago, IL)

Histotripsy is a focused ultrasound therapy that ablates tissue via the mechanical activity of bubble clouds. While effective for healthy tissues, histotripsy-induced bubble expansion will be mitigated in stiff chronic pathologies. In this study, the bubble cloud activity necessary for liquefaction of agarose phantoms with elastic moduli ranging from 12.3 $\pm$ 3.67 to 142 $\pm$ 44.9 kPa was investigated. Bubble clouds were initiated with 1-MHz pulses of 5-μs duration and peak negative pressures of 12 to 24 MPa. Bubble cloud emissions were mapped via passive cavitation imaging, and correlated with liquefaction using receiver operating characteristic analysis. The maximum power of emissions, and azimuthal location of the maximum power, were recorded for each experimental condition. For phantoms with elastic moduli between 12.3 and 85.8 kPa, no change was indicated in the bubble activity necessary for liquefaction. A larger acoustic power was associated with liquefaction of phantoms with elastic moduli of 142 kPa compared to 22.1 kPa. For a given peak negative pressure of the histotripsy pulse, no change in the peak power or azimuthal location of peak emissions was observed. These results indicate that a fixed bubble activity dose predicts histotripsy liquefaction over a wide range of medium stiffness.

Session 2pED

Education in Acoustics: Acoustics Education Prize Lecture

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

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

Chair's Introduction—3:25

Invited Paper

3:30

2pED1. The University of New Orleans ocean acoustics program at the Stennis Space Center, Mississippi. Stanley A. Chin-Bing (Dept. of Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, chin-bing@att.net)

In the late 1970's two US Navy ocean R&D organizations were formed and located at the Stennis Space Center (SSC), Mississippi, located approximately 50 miles East of the University of New Orleans (UNO). These two organizations employed nearly 1500 scientists, engineers, and technicians. Many of those with a Bachelor degree in physics, oceanography, or engineering desired advanced training in acoustics and signal processing. In 1982, George E. Ioup took the initiative to have UNO develop and teach acoustic courses on-site at the SSC. In the following 33 years, I developed 11 different graduate level courses in acoustics and taught them multiple times at the SSC. It was possible to take all the necessary courses needed for the Masters and PhD degrees on-site, while maintaining full-time employment. Several dozen Navy scientists received advanced degrees in physics with a specialty in acoustics from UNO. Many more received specific training in acoustics that enhanced their professional careers. This presentation will highlight my contribution to the UNO program at the SSC, and discuss how I incorporated my own research in ocean acoustic propagation and scattering into the many acoustic courses that I taught.

Session 2pID

Interdisciplinary and Education in Acoustics: Promoting Student Publishing Success

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

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

Rajka Smiljanic, Cochair
Linguistics, University of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198

Anders Lofqvist, Cochair
Haskins Labs., 300 George St., New Haven, CT 06511

Chair's Introduction—1:00

Invited Papers

1:05

2pID1. Promoting student publishing success: The progressive co-authorship cycle. Ann Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

I will describe the *"progressive co-authorship cycle"* as a strategy for promoting student publishing success. Crucial points in this progression are: (1) an initial phase in which the advisor is the lead author on a co-authored publication, (2) a middle phase in which the student and advisor are both in transition, and (3) a final phase in which the role reversal is complete and the student is the lead author on a co-authored publication. Each phase serves a unique purpose in the overall progression: modelling of good practice (phase 1), infusion of new energy and dynamism (phase 2), and finally, consolidation of a new and more equitable partnership (phase 3). I will emphasize that, as the fulcrum of the cycle, the middle phase is the most important yet easiest-to-neglect phase in a student-advisor relationship. During this phase, the student develops other co-authorship opportunities (e.g. with fellow students and/or other mentors). By infusing new ideas and research strategies into the mentor-mentee relationship, these "external" collaborations can be particularly enriching for phase 3. While this progression is rarely realized in clearly demarcated phases, it can provide a guiding framework for mentoring, and specifically for supporting student publishing success.

1:25

2pID2. Writing strategies for students and their mentors: From the perspective of a recent student. Kelly L. Whiteford (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu)

Writing an empirical paper can be a daunting process, particularly for students writing their first manuscript. This presentation will provide advice for helping students develop a structured and goal-oriented writing routine to aid in the publishing process. First, the mentor can help foster motivation to write by guiding the student to pursue research that is tethered to the student's intrinsic interests but still within the domain of the mentor's expertise. Second, writing goals need to be prioritized by the student in a specific, achievable, and time-bound manner. This prioritization can occur through creating a structured, weekly outline of writing-related goals, either with guidance from the mentor or through a peer support group. Setting aside regular, protected writing time can help form a habit out of writing and ensure it does not become pushed aside for other priorities. Lastly, -timely, thorough, and constructive feedback from the mentor can speed up the writing process while conveying to the student that their work is a priority of the mentor.

1:45

2pID3. Empowerment through POMA: How conference proceedings help students publish. Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

When it comes to publishing research results in a thesis or journal article, a disconnect between advisor expectations and student abilities and motivation can be a source of frustration, fear, and deteriorated relationships. Publishing in conference publications, including *Proceedings of Meetings on Acoustics* (POMA), represents an opportunity to bridge this gap. This presentation describes several benefits of publishing in conference proceedings, how to use POMA as a springboard, and some suggestions to help students and advisors overcome the barriers to publication.

2:05

2pID4. Perspectives on mentorship in the world of publishing. Ruth Litovsky (Commun. Sci. & Disord., Univ. of Wisconsin-Madison, 1500 Highland Ave., Waisman Ctr. Rm. 521, Madison, WI 53705, litovsky@waisman.wisc.edu)

Emerging scientists, students and postdoctoral fellows are typically being trained and are developing relationships with their mentors. This presentation will focus on issues that include: (1) timeline and goals related to one's career path; (2) strategic planning in publishing your work during the training period; (3) continuing to publish after moving on to other positions; (4) selecting journals and ethics in determining authorship; (5) how to get a mentor who is quite busy to respond to your drafts in a timely manner; (6) publishing quality vs. quantity; (7) reading and interpreting reviews; (8) getting involved in writing reviews when you advance in your career path. The presenter, Ruth Litovsky, received her PhD in 1991 and is Professor at the University of Wisconsin Madison, where she mentors students, post-docs and junior scientists.

2:25–2:40 Break

2:40–4:10 Panel Discussion

TUESDAY AFTERNOON, 14 MAY 2019 COMBS CHANDLER, 3:00 P.M. TO 5:30 P.M.

Session 2pMU

Musical Acoustics: Bluegrass Music and Related Instruments

Whitney L. Coyle, Chair
Department of Physics, Rollins College, 1000 Holt Ave, Winter Park, FL 32789

Chair's Introduction—3:00

Contributed Papers

3:05

2pMU1. On the phenomenon of natural-frequency splitting of a guitar string caused by a magnetic field. Jack Feinberg (Phys., Univ. of Southern California, Los Angeles, CA 90089-0484, feinberg@usc.du) and Bingen Yang (Aerosp. and Mech. Eng., Univ. of Southern California, Los Angeles, CA)

A magnet can affect the vibration of the metal strings in a musical instrument. We show that the magnetic field from a magnetic pickup can cause a frequency splitting of a metal guitar string's normally degenerate transverse vibration modes, leading to a beat note in the resulting sound. The amount of frequency splitting induced by the magnet depends on the product of the induced magnetization in the ferromagnetic string, and the magnet's spatial gradient at the position of the string, and is on the order of a few Hz. We apply free vibration theory to the string to obtain an eigenvalue problem, which we solve using a distributed-transfer function method. This method accurately predicts the natural frequencies of the vibrating guitar string in a non-uniform magnetic field. Videos of a high-E string vibrating with and without a nearby magnet will be shown.

3:20

2pMU2. The musical saw: Musical acoustics of trapped vibrational modes in a curved blade. Randy Worland (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

The musical saw is a popular folk instrument consisting of a flexible hand saw that is bent into an "*S*"-shape by the performer. When the smooth edge of the saw is bowed, skilled musicians can produce intricate melodies, as well as vibrato and glissando effects, by varying the curvature along the blade. The acoustically significant vibrational modes are trapped near the inflection point between regions of alternating curvature along the saw, exhibiting very low damping as a result of this confinement. Experimental data, taken using electronic speckle-pattern interferometry, illustrate the relevant mode frequencies, shapes, and positions relative to the blade curvature for various configurations of the saw. These data are presented and interpreted along with results from a finite element model of the tapered and curved saw blade.

Invited Papers

3:35

2pMU3. Investigations into the acoustics of the mandolin. Steve Tufte, Samuel Hunt, and Gerrick Hegarty (Dept. of Phys., Lewis & Clark College, MSC 15, 0615 SW Palatine Hill Rd., Portland, OR 97219, tufte@lclark.edu)

The mandolin is a descendent of the lute with a long history starting in 18th century Italy. This talk will give a brief history of the mandolin culminating in the modern flat-back mandolins used in bluegrass music. After a summary of previous research into its acoustics, we will describe our current research. This includes the use of Electronic Speckle Pattern Interferomentry (ESPI) to study the low-modes of the front and back plates and studies of the driving point impedance at the bridge. The mandolin has four courses of doubled strings tuned in unison. Inspired by Weinreich's seminal study of piano strings, we explore the coupling of the strings' motions to each other through their attachment to the vibrating bridge. By using a macro lens on a high-speed camera, we are able to study these motions in exquisite detail. We observe clear evidence of coupling including the exchange of energy between the vertical and horizontal motions of each string and correlations in the phases of their motions. We interpret the experimental results in the context of a theoretical model. [Work supported by the National Science Foundation under Grant No. 1707978.]

3:55–4:10 Break

Concert
4:10–5:30

Following the technical presentations will be a bluegrass music concert, all are welcome. The local bluegrass group RELIC will perform and discuss the history of the bluegrass genre and the instruments themselves.

Aaron Bibelhauser is a singer, songwriter, and instrumentalist from Louisville, Ky. In addition to writing songs recorded by award winning bluegrass artists including Balsm Range, Del McCoury Band, Michael Cleveland & Flamekeeper, and Dale Ann Bradley, he's taken first place in the Chris Austin Songwriting Contest as Merlefest and earned a nomination for the IBMA's prestigious Song of The Year Award. In addition to being an accomplished solo recording artist, radio broadcaster, and session player, Bibelhauser fronts the Louisville, KY based bluegrass band, Relic, along with his twin brother, Adam. Hailing from a city that bears a rich history of honoring the roots of traditional music while stretching boundaries and embracing progress. Relic is part of revival of this platform, blending rich vocal harmonies with colorful instrumentation.

Session 2pNS

Noise, Architectural Acoustics, ASA Committee on Standards, and Animal Bioacoustics: Soundscape and its Application Based on the New Standard

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

Klaus Genuit, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066

Chair's Introduction—1:30

Invited Papers

1:35

2pNS1. The soundscape standard—Its development and challenges. Brigitte Schulte-Fortkamp (TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

Soundscape entered the debate of noise annoyance issues in the nineties, most likely at the same time as when the EU Directive 2002/49/EC [PJ1] on Noise was under development. In 2014, the ISO standard ISO 12913-1: "Acoustics-Soundscape-Part 1: Definition and conceptual framework" was published, delivering the first framework for soundscape. This was followed in 2018 by "ISO/TS 12913-2 Acoustics—Soundscape—Part 2: Data collection and reporting requirements", which provided evaluation processes and information for integrating stakeholder participation. The third part of the standard is now underway: ISO/ TS 12913-3 "Acoustics- Soundscape—Part 3: Data analysis." Together, the international standard ISO 12913 provides guidance for accessing the key components in soundscape: people, acoustic environment, and context. The available standards in soundscape, psychoacoustics and noise management have provided a large step towards enhancing the quality of life for people. the urgent need for recognition but also for application with regard to new understandings of urbanism will be discussed.

1:55

2pNS2. Consequences in standardizing the soundscape data collection process. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

The ISO/TS 12913-2 dealing with data collection and reporting issues in the context of soundscape investigations was published in 2018. This technical specification proposes a range of methods and tools to perform the data collection process according to the current understanding of the soundscape approach. It intends to harmonize the data collection of soundscape studies and to ensure a certain level of comparability between soundscape investigations. In the past the vast majority of soundscape studies applied very different methods and tools limiting the comparability of results and avoiding the performance of meta-analyses. The reason might be the interdisciplinary roots of the soundscape concept asking for diverse disciplinary accesses to the phenomenon challenging the soundscape standardization efforts. Now, the disadvantage of the limited comparability of soundscape results might be overcome due to the publication of the ISO/ TS 12913-2. The paper will discuss the expected impact of the technical specification defining the data collection requirements on soundscape research and will highlight the benefits and potential drawbacks in standardizing the holistic soundscape approach.

2:15

2pNS3. Binaural measurement and psychoacoustic analysis—An advantage for the environmental noise research. Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

The new ISO TS 12913-2 "Data collection and reporting requirements" describes a lot of new aspects how to analyze soundscape (e.g., environmental noise) and provides new tools for the measurement and analysis. Whereas the use of binaural recording and psychoacoustic analysis is well known in the field of product sound quality especially with respect to the automotive field, the application of these tools within the soundscape analysis is new. This paper illustrates based on different examples given from the field of environmental noise the superiority and advantage of these tools in comparison to the conventional measurement using an omnidirectional microphone calculating the A-weighted sound pressure level.

2:35

2pNS4. An experimental soundscape study, combining binaural recordings, *in situ* questionnaires and behavioral mapping. Antonella Radicchi (Institut für Stadt- und Regionalplanung, Technische Universität Berlin, Hardenbergstraße 40 a, Sekr. B 4, Berlin, Berlin 10623, Germany, antonella.radicchi@tu-berlin.de)

In 2014 and, more recently, in 2018 the ISO norms on soundscape were released with the aim of providing a conceptual framework and standardized data collection methods for the international community of scientists and professionals with an interest on soundscape. This paper presents a research study conducted in the summer of 2018 in a Berlin public square where, following the ISO norms on soundscape, methods were applied to further investigate the findings of the "Beyond the Noise: Open Source Soundscapes" project, i.e., the everyday quiet areas as identified by the participants in the project. First, the paper introduces the research questions originated from the previous study. Second, it outlines the fieldwork procedure and methods, consisting in binaural measurements made in parallel to *in situ* questionnaires and behavioral mapping. Third, it discusses the initial results and limitations of the study. In conclusion it argues the soundscape approach potential within urban design and planning, illustrating how the results were partially exploited to design a bottom-up master plan proposal so as to improve the public square's soundscape and reduce noise pollution.

2:55–3:10 Break

3:10

2pNS5. Soundscape for smart tourism in Macao. W M To (Macao Polytechnic Inst., Macao, Macao) and Andy Chung (Macau Instituto de Acoústica, Macao, Macao, Macao, ac@smartcitymaker.com)

This paper presents a new initiative in enhancing the experience of tourists visiting Macao, by way of appling soundscape based on the new technical specification ISO/TS 12913-2:2018.

3:30

2pNS6. Prediction model of soundscape assessment based on semantic and acoustic/psychoacoustic factors. Ming Yang (HEAD Acoust. GmbH, Herzogenrath 52134, Germany, mingkateyang@163.com)

While conventional sound/noise environment assessment, management and relevant standards are based on the objective measurement of sound pressure level (SPL), soundscape research showed that humans' subjective assessment (such as positive and negative) cannot be well explained by the SPL or other acoustic indicators alone. Rather, soundscape assessment depends on many more factors such as cognition and context, which also need to be considered as suggested in the new soundscape standard. Among the different factors, a large number of previous studies supported that semantic factors, i.e. the sound sources which compose the sound environment—e.g., natural sounds (moving water, bird song, etc.), mechanical sounds (road traffic, construction, etc.) and human activity sounds, play the most important role in human perception and assessment. Therefore, from the systematic literature review of the relationship between such semantic factors and soundscape assessment, this present study proposes a model based on the sound source information, as well as SPL and psychoacoustic indicators of each of the different sound sources, to explain/predict largely, even though not fully, the soundscape assessment. Furthermore, for the practical use of this model which would be independent of manual efforts in recognition of sound sources through listening, this study discusses the possibility of integrating a computer algorithm of sound source recognition from field recordings of sound environment, for automated soundscape assessment.

3:50

2pNS7. Automated parsing of sound level meter data into artifacts, natural sources, and noise. Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

The National Park Service (NPS) has nearly one million hours of 1/3rd octave band sound level measurements collected at hundreds of sites throughout the system. One of the principal purposes of these data is to measure the background or residual sound level against which all transient sounds are heard. Historically this analysis has depended upon trained listeners to identify which sound sources can be heard in segments of these data. These annotated records were then used to calculate an adjusted median sound level. To pursue a more efficient, automated method to deliver this result, matrix decomposition methods were tested to characterize their capacity to model time series of sound level spectra as low-rank decompositions, including options to account for anomalous events. These methods were combined with sound source identification based on component spectral properties and time weightings to yield promising approaches for automating NPS analyses. These results may also offer options for detecting and removing the effects of pseudonoise due to the turbulence generated by air flowing past the microphone.

Contributed Paper

4:10

2pNS8. Evaluating manufacturing environment soundscapes. Brian J. Puckett and Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182, puckett.brian@huskers.unl.edu)

This work presents an investigation into the soundscape of a large, industrial manufacturing plant in Lincoln, Nebraska, including exploration of the noise reduction performance of sound enclosures. Occupational noise-induced hearing loss is among the most common nonfatal work-related injuries and illnesses in the U.S., and most efforts to reduce this exist as limited noise exposure times and mandatory use of hearing protection devices. This research aims to develop knowledge of factory soundscapes and explore acoustical strategies to reduce noise near the source. The factory in this study uses two distinct types of sound enclosures on site: rigid box-style enclosures and resilient curtain-style enclosures. Using state-of-the-art technology, it was possible to better understand contributions of single noise sources to the complex, overall soundscape and evaluate some detailed acoustical characteristics of the noise. The measured data are compared with subjective assessment of company employees using methods described in Part 2 of the Soundscape Standard (ISO/TS 12913-2). Results presented include a noise reduction comparison of the two types of enclosures, a relationship between noise reduction and source-enclosure proximity, and a comparison between measurements and survey data.

4:25–4:45 Panel Discussion

Session 2pPA

Physical Acoustics and Noise: Nonlinear Acoustics for Non-Specialists II

Won-Suk Ohm, Cochair
Yonsei University, 50 Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea

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

Chair's Introduction—1:30

Invited Papers

1:35

2pPA1. Curious nonlinearity of rocks. Carly M. Donahue and Paul A. Johnson (Earth and Environ. Sci., Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, cmd@lanl.gov)

Many geological materials, ranging from "rocks to unconsolidated sand," exhibit highly nonlinear elastic properties. Rocks fall in to a class of materials know as Nonlinear Mesoscopic Elastic Mesoscopic materials (NMEMs) in which the nonlinearity they possess is not derived from the constituent material, but rather the microscopic structure. Their behavior manifests as characteristic wave distortion, and slow dynamics, a recovery process to equilibrium that takes place over hours, days, weeks and sometimes years after a wave disturbance. A number of acoustic techniques have been developed to quantify a the material's nonlinear elastic coefficients and image localized damaged areas; and while much has been learned, much is left unknown, particularly identifying and understanding the underlying physical mechanisms that give rise to a rock's nonlinear elastic response, such as frictional losses, soft regions, and the influence of water content. But rocks also are a platform to understand other localized and distributed nonlinearity. Nondestructive evaluation techniques of metal and concrete are being developed with applications towards crack detection and wellbore integrity. Nonlinear acoustic techniques have recently appeared promising for performance evaluation of pressed powders and additively manufactured materials. In this presentation, we will provide an overview of nonlinear elasticity as illustrated by some examples.

1:55

2pPA2. Nonlinear acoustic metamaterials. Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Mark F. Hamilton, and Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

Acoustic metamaterials (AMM) have become a very active topic for research in numerous domains of engineering and science because of their promise to create materials, structures, and devices that can control acoustic wave propagation in ways that exceed the capabilities of naturally occurring or conventional composite materials. The majority of AMM research has been focused on linear behavior such as negative dynamic effective stiffness and density, cloaking, and negative refraction. One drawback of the focus on linear behavior is the restriction of the effective material properties of interest to narrow frequency bands that cannot be changed with external stimulus. Nonlinearity has been explored as a means of increasing the bandwidth of performance by enabling tunable band gaps and material configurability. Other recent work has investigated nonlinear AMM to access phenomena such as harmonic generation, non-reciprocity, enhanced energy absorption, solitons, mode hopping and conversion, chaos, and intrinsic localized modes. This talk will provide an overview of nonlinear AMM, starting with a background on AMM, then surveying existing research on the topic and its relationship to seminal works in nonlinear acoustics, and finally discussing promising avenues of future research. [Work supported by the National Science Foundation EFRI program and ARL:UT.]

2:15

2pPA3. Nonlinear acoustics of complex solids and granular media. Vincent Tournat (LAUM, UMR CNRS 6613, Le Mans Université, Av. O. Messiaen, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr)

This talk is an introduction to nonlinear acoustic processes in complex solids such as cracked metals, damaged composites or concrete, and granular media. These materials, belong to the same class of complex solids often referred to as mesoscopic solids, and share a number of similar nonlinear acoustic behaviors originating mainly from the presence of internal solid contacts. The specific nonlinear acoustic signatures and processes will be described (slow dynamics, memory, contact acoustic nonlinearity, i.e. nonclassical nonlinearities) through several examples, and an overview of the main modeling tools will be provided. Finally, the current trends and several applications of such nonclassical nonlinear processes (nondestructive testing and characterization, wave control, etc.) will be given.

2:35

2pPA4. Nonlinearity in high amplitude focused waves in solids for crack detection using time reversal techniques. Brian E. Anderson, Sarah M. Young (Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, bea@byu.edu), Marcel Remillieux (Geophys. Group, Los Alamos National Lab., Los Alamos, NM), Pierre-Yves Le Bas, and Timothy J. Ulrich (Detonation Sci. and Technol. Group, Los Alamos National Lab., Los Alamos, NM)

Defects in a sample cause localized distortions of a wave. Focused waves at a defect create amplitude-dependent signatures that may be quantified in post processing, thereby offering a means of locating and characterizing the defect. Time reversal (TR) is a technique that allows intentional focusing of wave energy. TR has been developed for crack and defect detection in solid media. Because TR provides a localized focus of elastic energy, it can be used to study the local nonlinear properties at point(s) of interest that are indicative of the presence of cracks and defects. Source transducers may remain in place as a sample is scanned for defects. A laser Doppler vibrometer offers a noncontact means of selecting points at which to focus energy. TR applied to detect cracking in steel rods will be presented. The results demonstrate that cracks may be identified through their nonlinear signatures when TR is used to focus energy at various positions along the rods.

2:55

2pPA5. On the utility of C/S as an indicator of nonlinearity in finite-amplitude dispersive waves. Taeyoung Park, Won-Suk Ohm (Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, ohm@yonsei.ac.kr), Kent L. Gee, and Brent O. Reichman (Brigham Young Univ., Provo, UT)

A frequency-domain nonlinearity indicator C/S, recently introduced by Ohm *et al.* [*Proc. Mtgs. Acoust.* **29**, 045003 (2016)], is a companion indicator to the well-known Morfey-Howell Q/S [*AIAA J.* **19**, 986–992 (1981)]. The two indicators can be regarded as "two sides of the same coin," because C/S and Q/S are, respectively, defined by the real and imaginary parts of the cross-spectrum of the pressure and squared-pressure waveforms. Despite the common origin, however, C/S and Q/S describe the pertinent nonlinear wave process in a different light. Furthermore, for nonlinear waves in media with strong dispersion only C/S may be applicable, because the use of Q/S could lead to erroneous interpretations. In this paper, we demonstrate the utility of C/S in the context of the Korteweg-de Vries (KdV) equation, which is one of the most celebrated model equations for nonlinear waves in dispersive media. The success of C/S is juxtaposed with the limitation of Q/S in describing nonlinear dispersive wave processes, especially in the case of solitons.

3:15–3:30 Break

Contributed Papers

3:30

2pPA6. Laboratory study of nonlinear propagation of transient signals in a sandstone bar. Thomas G. Muir, John M. Cormack, Charles M. Slack, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, P/O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu)

Experiments are reported on finite-amplitude, impact propagation in a thin bar of Texas moss sandstone. Wideband, unipolar pulses were generated by excitation with a pendulum hammer at one end of the bar, in contrast to prior work with either tone-burst excitation or resonance methods. The bar was rectangular, 8 cm on each side and 175 cm in length, with a density of 2.0 g/cc. Impacts had a center frequency around 3 kHz, amplitude of 10 to 130 microstrain, and propagated repetitively between reflections at each end. Measurements were made with a laser Doppler vibrometer at multiple locations along the bar, while ultrasonic tone-burst probes were used at a central location, transverse to the axis. The sound speed and effective modulus were reduced in proportion to the impact amplitude as the finite-amplitude impact pulse propagated through the transverse and other measurement sites. The attenuation of the impact signal was also found to follow a power law in the frequency range of 1–6 kHz, in proportion to impact intensity. Although amplitude-dependent attenuation and softening of the elastic modulus are both found in micro-inhomogeneous materials, the present results were enhanced, up to an order of magnitude, compared to those in previously reported experiments.

3:45

2pPA7. Modeling of plane progressive waves in media that exhibit slow dynamics and dissipative nonlinearity. John M. Cormack, Thomas G. Muir, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

Nonclassical nonlinearity refers to amplitude-dependent effects that are irreversible, for example slow dynamics and dissipative nonlinearity, both phenomena that are commonly observed in materials such as sandstone and concrete. Analyses of progressive waves in such media have been few, and have relied on hysteresis models that are difficult to implement, especially for transient or strongly nonlinear pulses. In this work, an evolution equation for plane progressive waves in a nonclassically nonlinear material is derived using two phenomenological models: the internal variable model of Berjamin *et al.* for slow dynamics [Proc. Roy. Soc. A **473**, 20170024 (2017)], and a simple model for dissipative nonlinearity [Zaitsev and Nazarov, *Acoust. Phys.* **44**, 362 (1998)]. The evolution equation is easily solved numerically given an arbitrary initial waveform. An initially narrowband signal exhibits pulse lengthening, amplitude-dependent attenuation, and nonclassical waveform steepening during propagation. Experimental observation of the same phenomena were reported recently by Remillieux *et al.* [*J. Geophys. Res.* **122**, 8892 (2017)]. Their results are reproduced in this study using the new evolution equation, showing good agreement between the model and the measurements, with additional insight into the behavior of the material gained from the model. [J.M.C. supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:00

2pPA8. Direct detection of dark matter particles by nonlinear effects in crystal vibrations. Igor Ostrovskii (Phys. and Astronomy, Univ. of Mississippi, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu)

The energetic dark matter particles (EDP) of $(4+1.5) \times 10^{15}$ eV mass-energy propagating from the central region of the Milky Way galaxy (CMW) are directly detected with the help of half-inch ~10 MHz quartz vibrators. The spectra of acousto-electric vibrations of thin piezoelectric crystal are theoretically calculated, and a nonlinear effect of external frictional influence is taken into account. The EDP produce the changes in the spectra of vibrations under specific orientation of the crystallographic axes toward the CMW. The mechanism of EDP-crystal interaction may be associated with the gravitational influence of EDP and their quantum properties, such as matter waves. The EDP average energy density is $0.71 + 0.29$ (GeV/cm^3) near Earth, which agrees with the expectations and existing astrophysical bounds of dark matter density in the solar neighborhood. The EDP can

transfer their momenta-energy to a media of propagation. The influences of EDP, or naviten, on vibrating atoms are detected from 10 different samples during 5 years. The micro-void tracks of EDP-particles are also observed in the vibrating plates of fuzzed quartz, which appear due to EDP decay inside solid media. The existence of EDP-particles is in agreement with recent detection of the ~10^{15} eV energy neutrinos from the CMW region.

4:15

2pPA9. Evolution equation for nonlinear Lucassen waves. Blake E. Simon, John M. Cormack, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, blakesimon8@utexas.edu)

A nonlinear, fractional, surface-wave equation was developed recently by Kappler *et al*. [*Phys. Rev. Fluids* **2**, 114804 (2017)] for propagation along an elastic interface coupled to a viscous incompressible medium. Linear theory for attenuation and dispersion of such a wave was developed originally by Lucassen [*Trans. Faraday Soc.* **64**, 2221 (1968)]. Kappler *et al*. employ a fractional derivative to account for the Lucassen attenuation and dispersion, and they include quadratic and cubic nonlinearity of the elastic interface. Presented here is a simplified form of their model equation for plane progressive waves. The resulting nonlinear evolution equation has the form of a Burgers equation but with a fractional derivative in place of the second derivative for viscosity, and with cubic as well as quadratic nonlinearity. In addition to facilitating analytical and numerical calculations, the evolution equation enables interpretation of a threshold phenomenon, revealed in numerical simulations presented by Kappler *et al*., as competition between quadratic and cubic nonlinearity. It is also suitable for determining critical source amplitudes above which Lucassen attenuation and dispersion alone cannot preclude formation of unphysical multivalued waveforms [Cormack and Hamilton, *Wave Motion* **85**, 18 (2019)]. [BES and JMC were supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:30

2pPA10. Nonlinear acoustic crack detection in thermoelectric wafers. John Greenhall, Alan Graham, and Cristian Pantea (Mater. Phys. and Applications, Los Alamos National Lab., MS D429, Los Alamos, NM 87544, jgreenhall@lanl.gov)

We present an acoustic technique for noninvasive crack detection in small (approximately 30.0 ×18.0 × 0.5 mm) thermoelectric wafers. The technique is based on exciting the wafers with a low-frequency signal that drives the crack to open and close periodically, and a high-frequency signal that is permitted to propagate through the closed crack and prohibited from propagating through the open crack. Interaction between the low- and high-frequency signals and the crack leads to generation of acoustic nonlinearities in the wafer. In contrast to existing acoustic crack detection techniques we utilize standing waves within the wafers to facilitate simultaneous crack detection throughout the wafer, we do not require uniform dimensions and material properties between wafers, and we do not affix the transducer to the wafers to avoid damaging the wafers. We present a mathematical model of the acoustic nonlinearity generation process and develop a procedure for identifying cracked wafers. We implement this technique experimentally and correctly identify cracked and crack-free wafers with total error in low single digits. This acoustic crack detection technique finds application in manufacturing of thermoelectric wafer and other wafers, where identifying cracks early in the manufacturing process results in significant time and cost savings.

4:45

2pPA11. Slow-dynamical nonlinearity in a glass bead pack probed with diffuse ultrasound and coda-wave interferometry. Richard Weaver, John Y. Yoritomo (Dept. of Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL 61801-3080, yoritom2@illinois.edu), John Popovics, and James Bittner (Dept. of Civil and Environ. Eng., Univ. of Illinois, Urbana, IL)

Low amplitude diffuse coda ultrasound probe signals received through unconsolidated glass bead packs under static vertical loads are used to measure the bead pack's slow dynamics, in which the speed of low amplitude probe waves varies slowly with time long after loading. Coda-wave interferometry reveals tiny stretches in the waveform, with precision of parts in 10^5. These stretches are ascribed to decreased contact stiffnesses or decreased sound speeds. The coda-wave stretch is observed to be sensitive to low frequency conditionings at strains of order 10^{-5}. Such conditionings include taps on the sides, low frequency harmonic vibrations and impulses on top, and step changes in vertical load. Typical behavior after the conditioning is an initial loss of stiffness followed by a slow healing that proceeds with the logarithm of time. The initial loss of stiffness and its slow dynamic healing are investigated for sensitivity to sundry parameters.

Session 2pPPa

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Elizabeth A. Strickland, Chair
Speech, Language, and Hearing Sciences, Purdue University, 500 Oval Drive, West Lafayette, IN 47907

Chair's Introduction—1:00

Invited Paper

1:05

2pPPa1. Differences and similarities of peripheral auditory systems. Glenis R. Long (Speech-Language-Hearing Sci., Graduate Ctr. CUNY, 365 Fifth Ave., New York, NY 10016, glong@gc.cuny.edu)

The properties of the ears of the wide range of species processing auditory signals can vary greatly depending on both important environmental sounds, and the sounds generated by individual members of the species (which are shaped by the sound processing and sound generation systems). Unfortunately, most anatomical and physiological research is conducted on a small subset of mammals and there are many pressures on auditory researchers to limit their research to species that are easy to work with in laboratories which are motivated to explain normal and impaired hearing in humans. Psychoacoustics has played a major role in our understanding of auditory processing in humans, but is difficult and time consuming in nonhuman species. Much of the investigation in nonhuman species is invasive and not appropriate for use with humans. Otoacoustic Emissions (sounds generated by the inner ear and measured in the outer or middle ears) can be used in most species, but we are still investigating the extent to which OAE obtained from different species reflect similar underlying processes. The combination of OAE measurement and computer modelling of the data has made us aware of the importance of evaluation of cross-species and individual differences in auditory processing.

Session 2pPPb

Psychological and Physiological Acoustics and Education in Acoustics: Cultivating New Growth by Composting Old Ideas: Pruning the Deadwood from the Garden of Psychological and Physiological Acoustics

G. Christopher Stecker, Chair
Hearing and Speech Sciences, Vanderbilt University, 1215 21st Ave. South, Room 8310, Nashville, TN 37232

Chair's Introduction—2:10

Invited Papers

2:15

2pPPb1. Critical examination of critical bands. Laurel H. Carney (Biomedical Eng. & Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu)

The concept of the critical band has driven the design and interpretation of psychophysical experiments for decades. The critical band was initially proposed by Fletcher [*Rev. Mod. Phys.* **12**, 47 (1940)] to describe results of masking experiments, and it is generally interpreted as evidence for a filter that limits the signal-to-noise ratio in stimulus representations that are provided to the central nervous system. This filter-based interpretation of masked-detection thresholds has been extended and applied to numerous tasks, including many without masking. The concept of a bank of critical-band filters also forms the basis for signal-processing strategies used for hearing aids and cochlear implants. This talk will review both psychophysical and physiological results that challenge the concept of the critical-band filter. For example, the minimal effect of roving-level paradigms on masked-detection thresholds directly refutes the critical-band based power-spectrum model. Furthermore, most auditory neurons have receptive fields that are much broader than critical bands at the moderate to high sound levels used in most psychophysical tasks, with or without masking. Alternative concepts that are robust across a wide range of levels and in roving-level paradigms, such as fluctuation profiles, explain the original masked-detection results and related experiments, such as psychophysical tuning curves.

2:40

2pPPb2. Challenging standard practices in adaptive psychophysics. Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, ehoover@umd.edu), Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., Portland, OR), and David A. Eddins (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

One of the most cited manuscripts in behavioral hearing research is Levitt [*J. Acoust. Soc. Am.* **49**, 467–477. (1971)]; it is typically the only reference cited in the procedures section of studies reporting thresholds obtained via adaptive tracking. This is problematic because Levitt (1971) informs only one among many parameters used in threshold estimation, and the relationship between Levitt's transformed up-down method and the proportion correct targeted by the procedure is commonly disrupted by the selection of the remaining parameters [*Vis. Res.* **38**, 1861–1881. (1998)]. We explored the evidence supporting standard practices and found that step size, stopping criterion, and threshold estimation from track data have limited theoretical motivation and can introduce systematic bias and error into threshold estimates as shown in behavioral and simulation studies. As a result, we propose an approach to developing and describing methods for behavioral threshold estimation that offer improved efficiency, reliability, reproducibility, and are no less consistent with Levitt (1971). [Work supported by NIH NIDCD DC015051.]

3:05

2pPPb3. Accounting for task switching when measuring listening effort: A cautionary tale for the dual-task paradigm. Adrian K. C. Lee (Speech and Hearing Sci., Univ. of Washington, Box 357988, Seattle, WA 98195, akclee@uw.edu)

Listening in everyday environments inevitably involves task switching. For example, one might be typing up an abstract while listening to children singing in the background or one might be listening to a radio news program while finding a parking space on a crowded street. While there is an extensive body of research that examines how cognitive control processes support the ability to adjust behavior dynamically, how task-switching impacts listening is less explored. In this talk, different kinds of costs associated with task switching will be reviewed. Specifically, how these well-studied switch costs could impact our interpretations of dual-task paradigm findings—an experimental framework that is often used to interrogate listening effort—will also be discussed. [Work supported by NIH R01 DC013260].

3:30

2pPPb4. Circling back on theories of sound localization. Antje Ihlefeld, Nima Alamatsaz (Biomedical Eng., New Jersey Inst. of Technol., 323 Martin Luther King Blvd., Fenster Hall, Rm. 645, Newark, NJ 07102, ihlefeld@njit.edu), and Robert M. Shapley (Ctr. for Neural Sci., New York Univ., New York, NY)

An important question of human perception is how we localize target objects in space. Through our eyes and skin, activation patterns on the sensory epithelium suffice to cue us about a target's location. However, for our ears, the brain has to compute where a sound source is located. One important cue for computing sound direction is the time difference in arrival of acoustic energy reaching each ear, the interaural time difference (ITD). With behavioral experiments on sound lateralization as a function of sound intensity, we tested how the computation of sound location with ITDs is done. We tested twelve naïve normal-hearing listeners (ages 18–27, five females). Stimuli consisted of low-frequency noise tokens that were bandlimited from 300 to 122 Hz, from 5 to 25 dB sensation level. Without response feedback, listeners were initially trained to reliably judge the direction of a sound source and then tested on where they heard the sound. We found that softer sounds tend to be localized closer to midline as compared to louder sound. This finding raises doubt on one major theory of sound localization, the labeled-line theory, and supports another main contender, population rate based coding.

3:55–4:10 Break

4:10

2pPPb5. "Straightness" versus "briefness" in binaural cue extraction. G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu), Mathias Dietz (Universität Oldenburg, Oldenburg, Germany), and Richard M. Stern (Carnegie Mellon Univ., Pittsburgh, PA)

The neural mechanisms that detect and encode interaural delays have elicited intense interest for many decades, from Jeffress's [*J. Comput. Physiol. Psychol.* **41**, 35–39 (1948)] original "place theory of sound localization," its modeling via frequency-specific cross-correlation starting with Colburn [*JASA* **54**, 1458–1470 (1973)], and continuing to this day. An important advance was the emphasis of cross-correlation peaks that align across frequency ["straightness" ; Stern *et al*., *JASA* **84**, 156–165 (1988)]. While narrowband cross-correlation produces multiple peaks and is therefore inherently ambiguous, straightness detection accounts for resolution of that ambiguity in broadband stimuli by combining binaural outputs across frequency. An alternative account is suggested by the effects of temporal fluctuations, such as onsets, on the *inputs* to binaural processing. Zurek [*JASA* **67**, 953–964 (1980)] described how emphasizing such events (e.g., by "windowing" the input) also resolves interaural ambiguities. Recent psychophysical and physiological evidence supports that view, strongly suggesting that binaural processing does, in fact, occur in brief windows triggered by envelope fluctuations such as onsets and intrinsic fluctuations in bands of noise. This talk investigates the possibility that the resulting temporal sparsity of binaural inputs ("briefness") might account for bandwidth effects even within single frequency channels, i.e. without a second level of "straightness" detection.

4:35

2pPPb6. What studies of audio-visual integration do not teach us about audio-visual integration. Ross K. Maddox (Biomedical Eng. and Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rm. 5.7425, Rochester, NY 14642, ross.maddox@rochester.edu)

Auditory perception depends on more than just the processing of acoustic stimuli. Visual stimuli can also have a profound influence on listening. Salient examples of such effects include spatial ventriloquism—in which the location percept of an auditory stimulus is "captured" by that of a simultaneous visual stimulus—as well as drastically improved understanding of speech in noise when the talker's face is visible to the listener. These phenomena are typically described as "audio-visual integration," and are often well modeled, as in the case of the ventriloquist effect, by ideal Bayesian causal inference. However, there is an over-reliance in these studies on single pairs of stimuli (i.e., one auditory and one visual stimulus) and the nature in which cross-modal discrepancies are resolved. This talk will first discuss two problems resulting from that: first, there is ambiguity about whether the integration occurs from weighing two independent sensory estimates or a single bound percept, and second, the design is less useful for studying integration when the stimuli are congruent. The talk will then describe recent work from our lab focused on new designs using multiple stimuli in an attempt to alleviate these issues and inform better models of integration.

Contributed Papers

5:00

2pPPb7. Revisiting perceived intracranial lateralization for stimuli with interaural time differences that are larger than the head. Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, 0119E Lefrak Hall, College Park, MD 20742, goupell@umd.edu), Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), Julia Nothaft, and H. Steven Colburn (Dept. of Biomedical Eng., Boston Univ., Boston, MA)

The interaural time difference (ITD) is the primary sound-localization cue for humans. While realistic sound sources have energy across a wide frequency range, the ear performs a narrowband frequency decomposition, and within each band are ambiguities in the interaural cross-correlation (an index of the signal ITD). These ambiguities are thought to be resolved by the approximate consistency of ITDs across frequency, or "straightness." However, straightness has not been evaluated over a wide range of stimuli. Therefore, normal-hearing listeners reported the intracranial lateralization of narrowband noises and tone complexes to better evaluate across-frequency ITD processing. In a new modification of the typical paradigm, listeners were encouraged to give multiple responses if split images were perceived. ITDs larger than those naturally produced by the head (~750 ms) best demonstrate straightness because across-frequency comparison is necessary to resolve the interaural phase ambiguities, which is why ITDs as large as 1500 ms were applied. Straightness effects reported previously for narrowband noises were replicated, and were broadly similar for complex tones. The extent of lateralization of 1500-ms ITDs and the occurrence of split images were difficult to account for using simple lateralization-based models for smaller ITDs.

5:15

2pPPb8. Flaws in the use of spectral ripples in cochlear implants. Matthew Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, 1417 NE 42nd St., Seattle, Washington 98105, mwinn83@gmail.com) and Gabrielle O'Brien (Speech & Hearing Sci., Univ. of Washington, Seattle, WA)

Spectral ripple discrimination is a popular measure of spectral resolution that has been shown to correlate with speech recognition scores in cochlear implant (CI) listeners. In the test, listeners distinguish sounds with varying density of spectral peaks, with some spectral modulation depth. We argue that there are numerous significant flaws with the application of the test specifically in CI listeners. To start, the spectrum is aliased by the CI processor in a way that is similar to frequency aliasing for under-sampled time series. Beyond a critical spectral density, the spectral envelope changes in a chaotic fashion and is no longer under experimenter control. This critical density is exceeded in numerous published studies. Furthermore, the densities linked with "good" performance are not only outliers, but are entirely unrelated to the spectral densities of real speech sounds, and likely exhibit undue leverage over correlation values. Additionally, there are reports of experience and learning effects, inconsistent with the often-stated goals of the test to avoid such factors. We show how artefactual nonlinearities at high spectral densities may unintentionally match the spectral envelope characteristics of speech sounds—an unfortunate result that likely has given spurious results that sustain the use of this test.

TUESDAY AFTERNOON, 14 MAY 2019 STOPHER, 1:15 P.M. TO 4:40 P.M.

Session 2pSA

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, Noise, and Architectural Acoustics: Acoustic Metamaterials II

Christina J. Naify, Cochair
Acoustics, Jet Propulsion Lab, 4800 Oak Grove Dr, Pasadena, CA 91109

Alexey S. Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, West Bethesda, MD 20817

Invited Papers

1:15

2pSA1. Plate mechanical metamaterials and their applications. Igor Bargatin (Mech. Eng. and Appl. Mech., Univ. of Pennsylvania, 220 South 33rd St., Philadelphia, PA 19104, bargatin@gmail.com)

Recently, we introduced the concept of *plate mechanical metamaterials*—cellular plates with carefully controlled periodic geometry and unique mechanical properties—as well as its initial realization in the form of freestanding corrugated plates made out of an ultrathin film. We used atomic layer deposition (ALD) and microfabrication techniques to make robust plates out of a single continuous ALD layer with cm-scale lateral dimensions and thicknesses between 25 and 100 nm, creating the thinnest freestanding plates that can be picked up by hand. We also fabricated and characterized *nanocardboard*—plate metamaterials made from multiple layers of nanoscale thickness, whose geometry and properties are reminiscent of honeycomb sandwich plates or corrugated paper cardboard. Ultralow weight, mechanical robustness, thermal insulation, as well as chemical and thermal stability of alumina make plate metamaterials attractive for numerous applications, including structural elements in flying microrobots and interstellar light sails, high-temperature thermal insulation in energy converters, photophoretic levitation, as well as ultrathin sensors and resonators. I will briefly discuss our experimental progress on all these applications, including demonstrations of extremely robust thermal insulators that can sustain a temperature difference of ~1000 °C across a micron-scale gap, macroscopic plates that levitate when illuminated by light, and hollow AFM cantilevers that offer greatly enhanced sensitivity and data acquisition rates.

1:35

2pSA2. Effects of geometry and mass distribution in 3D printed metastructures for vibration mitigation. Ignacio Arretche, Ganesh U. Patil, and Kathryn H. Matlack (Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 1206 W. Green St., Urbana, IL 61801, kmatlack@illinois.edu)

This presentation will discuss our recent progress on mechanical metastructures, which have engineered micro- and meso-scale features and are designed to have prescribed vibration mitigation properties. The goal of this work is to understand how the geometry and structure of a material can manipulate elastic wave propagation. Specifically, we developed a class of 3D printed metastructures that support low and wide band gaps, by combining periodic lattices with embedded resonant inclusions. The band gaps in these metastructures have significant shifts in frequency with small geometry changes in the unit cell design. We use finite element method simulations to calculate how band gaps in infinitely periodic metastructures depend on how the material is distributed throughout the unit cell. We study different lattice geometries that exhibit a variety of bend- and stretch-dominated deformations, as well as different volume fraction and

placement of resonant inclusions. We fabricate metastructures with 3D printing and measure their frequency-dependent vibration transmission to experimentally validate their behavior. These metastructures have engineering applications in structural components that prevent the propagation of damaging structural vibrations.

Contributed Papers

1:55

2pSA3. Randomness in periodic metamaterials. Hussein Esfahlani and Andrea Alu (Photonics Initiative, Adv. Sci. Res. Ctr., 85 St. Nicholas Terrace, The Towers at CCNY, Rm. 1G16, New York, NY 10031, hsesfahlan@gmail.com)

The most straightforward way to design artificial structures is to use periodicity and leverage its potential mathematical and physical consequences to our favor. However, strict periodicity, which is usually employed in designing artificial materials, is not practically feasible and most often some degrees of randomness are introduced in the fabrication process, which degrade the performance. In this presentation, using the Furstenberg theory on the product of random matrices and Monte Carlo simulations we explore the effect of randomness in the transmission performance of periodic structures designed to realize density-near-zero (DNZ) metamaterials. We show that DNZ propagation is very sensitive to randomness, and we unravel the physics behind this sensitivity with analytical theory and full-wave simulations.

2:10

2pSA4. An acoustic power divider with uniform phase and compressibility-near-zero effective material properties. Matthew S. Byrne (Elec. and Comput. Eng., Univ. of Texas at Austin, 1616 Guadalupe St., Austin, TX 78751, mbyrne@utexas.edu), Hussein Esfahlani (Photonics Initiative, Adv. Sci. Res. Ctr., New York, NY), and Andrea Alu (Photonics Initiative, Adv. Sci. Res. Ctr., Austin, Texas)

Materials with properties which can be described by a near-zero index have received much attention in the fields of microwave electromagnetics, optics, and acoustics, due to their extraordinary capabilities in wave manipulation. It was recently demonstrated, theoretically and experimentally, that acoustic media can support near-zero-index propagation, in which the effective compressibility of a waveguide channel approaches zero. In principle, this allows the complete tunneling of acoustic waves with nearly infinite wavelength (or equivalently, uniform phase). In this work, we show that these concepts can be extended to realize a novel acoustic power divider, which permits the tunneling of acoustic power to an arbitrary number of output ports, where the phase shift with respect to the input signal can be selected to be either 0 or 180 degrees. Analytical and numerical models which describe the behavior of the power divider, are presented. We conclude with an analysis that describes the limitations and trade-offs that occur due to losses as the device size is scaled.

2:25

2pSA5. Surface acoustic waves over bianisotropic metasurfaces. Li Quan (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 1616 Guadalupe St., UTA 7.215, Austin, TX 78712, liquan@utexas.edu) and Andrea Alù (Photonics Initiative, Adv. Sci. Res. Ctr., City Univ. of New York, New York, NY)

Metasurfaces are the two-dimensional planarized analogues of metamaterials, yielding unique sound manipulation along a surface. Our group recently proposed the concept of bianisotropic metasurfaces achieved by exploiting the strong nonlocal coupling between neighboring vibrating units, yielding anisotropic impedance in airborne acoustics, and we have applied this property to realize acoustic hyperbolic metasurfaces, which require extreme impedance anisotropy. In this talk, we investigate the wave propagation properties of bianisotropic metasurfaces and report all-angle backward-wave propagation on the surface. By carefully designing the coupling between unit cells, both negative phase index and energy index for surface acoustic waves are observed above the metasurface for all propagation directions, hence realizing converging waves for a radiating point source. Since in bianisotropic metasurfaces the energy can be transmitted not only through surface waves, but also through the coupling units, negative energy index for surface waves does not violate energy conservation laws, as it can be sustained by a positive energy flow within the connected unit cells composing the metasurface.

2:40–2:55 Break

2:55

2pSA6. Acoustic pressure enhancing metamaterials through impedance contrast. Hyung-Suk Kwon and Bogdan Ioan Popa (Dept. of Mech. Eng., Univ. of Michigan, University of Michigan, 2350 Hayward St, Ann Arbor, MI 48109, kwonhs@umich.edu)

Acoustic pressure enhancing devices are widely used to extend the range of sensors used for detecting weak and distant sounds. However, conventional sound enhancing devices such as acoustic horns and parabolic reflectors require bulky structures and thus limit miniaturization of sensing systems. To overcome this limitation, metamaterial techniques have been employed and promising results have been reported. However, acoustic pressure enhancing metamaterials reported so far rely on frequency dependent mechanisms to increase acoustic pressure such as resonances or wave compression methods. Therefore, in these approaches, waves are distorted during the enhancement process and this limits their applications considerably. In this presentation, we will show that metamaterials can significantly enhance acoustic waves without wave distortion while keeping the size of the metamaterial subwavelength. Our metamaterial is based on the property of the acoustic waves to increase their acoustic pressure while propagating without insertion loss from a medium of low impedance into a medium of higher impedance. The pressure gain is constant regardless of the frequency, allowing the wave to maintain its shape during enhancement. Here, we will provide the physics of the phenomenon along with numerical and experimental results which were in good agreement with the theoretical prediction.

3:10

2pSA7. Fabrication and measurement of a hybrid resonant gradient index metasurface at 40kHz. Nikhil J. Gerard (Mech. and Aerosp. Eng., North Carolina State Univ., 3141, 911 Oval Dr., EB3, Raleigh, NC 27606, rjoseph3@ncsu.edu), Huachen Cui (Mech. Eng., Virginia Tech, Blacksburg, VA), Chen Shen, Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC), Xiaoyu Zheng (Mech. Eng., Virginia Tech, Blacksburg, VA), and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

Over the past decade, gradient index metasurfaces(GIMs) have been voraciously studied for the numerous wave control capabilities that they facilitate. In this regard, a hybrid structure consisting of shunted Helmholtz resonators and a straight channel is often chosen as building blocks of the metasurface. Prior research, however, has primarily focused on GIMs that operate in the audible frequency range, due to the difficulties in fabricating such intricate structures at the millimeter and sub-millimeter scales, for ultrasonic applications. In this paper, we design, fabricate and experimentally realize a gradient index metasurface for airborne ultrasound at 40kHz. The fabrication of such a GIM is made possible by projection micro-stereolithography, an emerging additive manufacturing technique capable of microscale, high aspect-ratio features over a wide area. Simulations were first conducted to verify the metasurface design. Experiments were subsequently performed to corroborate the simulations and theory. The challenges faced by the thermoviscous effects, their usefulness in certain applications and optimal designs for minimal dissipation are discussed.

3:25

2pSA8. Synchronization and avalanching amongst coupled phase oscillators, mechanical lasing. John P. Coleman and Richard Weaver (Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL 61801, jpcolem2@illinois.edu)

We present theory and numerical simulations for two nonlinear systems consisting of a large number of eccentrically-weighted DC motors on mechanical bases. These are generalizations of the Kuramoto model for synchronizing phase oscillators, one chief difference being that the coupling is now frequency dependent. In one system the base is resonant and has a single degree of freedom. This system exhibits the expected second order phase transformation: for sufficient coupling strength the motors synchronize with a power output that grows with the distance above criticality. In the synchronized state the base oscillates at a single frequency (below its nominal resonance) with an amplitude that rises superluminescently with the number of motors. The degree of synchronization fluctuates intermittently, with statistics similar to those of universal crackling noise and avalanches. In our other system the motors are placed densely on a tensioned membrane, with sound speed such that wavelength is large compared to motor size and spacing. As such the structure is an active nonlinear metamaterial. As a function of coupling strength we observe a lasing transition from a near-quiescent state to a state in which the membrane is dominated by a single wavelength, and acoustic power emission is high.

3:40

2pSA9. Diffuse ultrasonic transmission between two half spaces coupled through a single glass bead. Richard Weaver and John Y. Yoritomo (Dept. of Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL 61801-3080, yoritom2@illinois.edu)

We consider ultrasonics in glass blocks in contact through a single 3 mm glass bead held in place by contact forces of up to five Newtons. Hertzian contact theory predicts resonant transmission at a few isolated frequencies between 65 and 120 kHz. Resonances, based on calculations of radiative losses to the blocks, are predicted to be narrow, with widths of order 50Hz. Wide-band diffuse ultrasound from 50 to 800 kHz launched by an impulse in the upper block leads to a diffuse signal in the upper block that slowly diminishes due to absorption. It leads to a diffuse signal in the lower block that slowly increases in amplitude due to transmission—through the air and through the bead—before dissipating due to absorption. The spectrum in the lower block includes a broadband part demonstratable as due to transmission through the airgap. It also includes a part due to transmission through the bead and confined to a few isolated frequency bands. These transmission bands have widths of several kHz. We also investigate slow dynamics at the contact points by studying the change and recovery of the diffuse transmission after large amplitude conditionings.

3:55

2pSA10. A comparison study between topological insulators based on valley Hall and quantum spin Hall effects. Yuanchen Deng and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27606, ydeng5@ncsu.edu)

Over the past few years, the rapid development in the fields of condensed matter physics, electronic and photonic systems have inspired the design and experimental demonstration of various acoustic topological insulators. Among these, the topologically protected one-way propagation is a phenomenon that is gaining increased attention. Pseudospin states, which is the analogue of the Quantum Spin Hall Effect from electronic systems, has been proven to enable topological edge states in acoustics. Similarly, Valley Hall (VH) effect is also observed in acoustic systems and provides a pair of valley vortex states with opposite chirality. These valley vortex states can similarly form topologically protected edge states and in turn realize robust one-way propagation. However, the differences in the physics behind these acoustic systems give rise to distinct features such as angle selection and immunization level to various types of defects. In this article, the comparison between topological insulators (TI) and valley hall topological insulators (VHTI) address the difference and similarities in several aspects. Both of them have topologically protected bandgaps and thus the robust one-way propagation. For the maximum transmission incident angle and defect immunization, however, VH topological waveguide and TI waveguide show different characteristics.

4:10

2pSA11. Transmission enhancement through arbitrary layers with bianisotropic metasurfaces. Junfei Li, Chen Shen, and Steven Cummer (ECE, Duke Univ., 101 Sci. Dr., Rm. 3417, FCIEMAS Bldg., Durham, NC 27708, junfei.li@duke.edu)

Acoustic metasurfaces are thin, engineered structures that can control the local reflection and transmission coefficients of acoustic waves. Here, we describe our recent framework on general impedance matching with bianisotropic metasurfaces, in which elements are designed to control the local asymmetric impedance matrix. Compared with conventional quarter wavelength impedance matching layers, our proposed metasurface, with arbitrary thickness, can maximize wave transmission through an arbitrary layer with controlled phase. As a showcase, we demonstrate the design of a metasurface that helps ultrasound in water to penetrate a steel plate with almost 100% power efficiency and controlled transmission phase, which is beyond what is possible with conventional impedance matching layers. Our approach also applies to electromagnetic scenarios to help waves transmitting through walls or even metal plates.

Invited Paper

4:25

2pSA12. Flat acoustics with soft gradient-index metasurfaces. Olivier Mondain-Monval, Raj Kumar (CNRS-CRPP, Univ. of Bordeaux, 115 Av Schweitzer, Pessac 33600, France, mondain@crpp-bordeaux.cnrs.fr), Thomas Brunet (Inst. of Mech. Eng., Talence, France), Yabin Jin (Inst. of Mech. Eng., Bordeaux, France), and Olivier Poncelet (Inst. of Mech. Eng., TALENCE, France)

There is an important demand for the manufacture of materials allowing a precise control of the propagation of acoustic waves. This depends, among other things, on the speed of propagation of the waves in the material, which is expressed by the so-called acoustic index: $n = C_{L,ref} / C_{L,mat}$ where $C_{L,ref}$ and $C_{L,mat}$ are the propagation velocities of acoustic waves in a reference medium (water for us) and in the material, respectively. We have recently shown that silicone-based porous polymer materials possess exceptionally high and scalable propagation indices, which can be tuned by a precise control of the polymer's mechanical properties and porosity [1]. Our materials are obtained using an emulsion templating method coupled with a drying process that will be described in my presentation. Using this experimental approach, we were able to produce materials with an acoustic index gradient allowing, among other things, the development of a flat acoustic lens [2]. I will present the synthesis and the characterization of these materials as well as their acoustic properties. [1] A. Kovalenko, M. Fauquignon, T. Brunet, and O. Mondain-Monval, "Tuning the sound speed in macroporous polymers with hard or soft matrix," *Soft Matter* **13**, 4526–4532 (2017). [2] Y. Jin, R. Kumar, O. Poncelet, O. Mondain-Monval, and T. Brunet, "Flat acoustics with soft gradient-index metasurfaces", *Nat. Commun.* (to be published).

Session 2pSC

Speech Communication and Psychological and Physiological Acoustics: Perception and Production of Speech Directed Toward Infants and Children (Poster Session)

Mark VanDam, Cochair
Speech & Hearing Sciences, Washington State University, P.O. BOX 1495, Spokane, WA 99202

Laura Dilley, Cochair
Department of Communicative Sciences, Michigan State University, East Lansing, MI 48824

All posters will be on display from 1:30 p.m. to 4:30 p.m. To give contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m.

Contributed Papers

2pSC1. Quantifying child directed speech cross-culturally across development. Melanie Soderstrom (Psych., Univ. of Manitoba, 190 Dysart Rd., Winnipeg, MB R3T 2N2, Canada, m_soderstrom@umanitoba.ca), Marisa Casillas (Max Planck Inst. for PsychoLinguist, Nijmegen, The Netherlands), Elika Bergelson (Duke Univ., Durham, NC), Jessica Kirby (Psych., Univ. of Manitoba, Winnipeg, MB, Canada), Celia Rosemberg, Alejandra Stein (CIIPME Conicet, Buenos Aires, Argentina), Anne Warlaumont (UCLA, Los Angeles, CA), and John Bunce (Psych., Univ. of Manitoba, Winnipeg, MB, Canada)

Child-directed speech (CDS) influences language development (e.g., Golinkoff *et al*., 2015), but varies across cultural and demographic groups (Hoff, 2006). Recent work examining speech heard by North American English (NAE) infants found an increased *proportion* of CDS with age (Bergelson *et al*., 2018). Quantity of CDS remained relatively constant across age, while quantity of adult-directed speech (ADS) decreased. We replicate these findings using a different methodology, and expand them to include other language communities. Our data come from daylong audio recordings of 58 children ages 2–36 months from the ACLEW dataset (Bergelson *et al*., 2017; 30 children acquiring NAE, 10 UK English, 8 Argentinian Spanish, and 10 Tseltal/Mayan). Ten randomly selected 2-min segments (Tseltal: nine 5-min segments) from each child were annotated for speaker gender, age (child or adult), and addressee for each utterance. We calculated the minutes per hour of CDS, ADS, and all speech. Preliminary analyses find high variability in overall language input across individuals, age, and culture, and partially replicate the Bergelson *et al*. (2018) pattern of results. Ongoing annotation will permit finer-grained analyses of sub-group differences. Further analyses will examine the influence of factors such as speaker gender, number of speakers, and maternal education.

2pSC2. Longitudinal development of conversational exchanges in children with hearing loss. Mark VanDam and Rebecca Hibben (Speech & Hearing Sci., Washington State Univ., P.O. Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu)

Conversations between adults and children have been shown to be an important factor affecting the development of cognition, academic performance, and language use in children. Conversational exchange frequency between parents and children has been shown to increase with age. It has also been demonstrated that degraded sensory input, such as with hearing loss, can result in reduced language, speech, and communication skills in children. This study examines the longitudinal development of conversational exchanges between children with hearing loss and their parents. Thirty-nine families contributed 370 daylong recordings constituting 4337 h of audio collected from a body-worn audio recorder. Automatic speech processing techniques were used to identify and tally conversational exchanges. The main longitudinal effect of age on conversation exchange rate is confirmed, but the effect is weaker and may have a different developmental trajectory for families of children with a hearing loss. This result may be influenced by differences in the development of joint attention between children who are typically developing and those with hearing loss.

2pSC3. Speech perception in children with reading disabilities: Phonetic perception is the problem. Yashuo Wu (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 512 E Green St. Apt. 301, Champaign, IL 61820, yashuow2@illinois.edu), Jie Lu (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Zhanjiang, China), Joseph C. Toscano (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), Cynthia Johnson (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Champaign, IL), and Jont b. Allen (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Reading disability (RD) is widely accepted as a key obstacle in the development of literacy. Studies show that 15–20% of grade-school students are RD. Many quit high-school and go to jail. We shall show that RD for 8–12 yrs is related to inadequate phonetic identification ability, rooted in preschool language development. We used two tests (10 thousand responses/child): (1) A 3-interval forced choice procedure (Syllable Confusion Oddball Task: SCO). (2) A single CV/VC presentation task with oral response, to label CV/VC phones (Nonsense Syllable Confusion Matrix: NSCM). The experimental results showed that for the SCO task the 10 RD cohort had, on average 5 times the error compared to the 6 RC reading control (RC) cohort. The errors were highly idiosyncratic, analyzed by logit. (1) RDs have significant speech perception problems, despite normal pure-tone hearing and language ability. (2) When comparing the SCO and NSCM results, our findings are consistent with a reduced ability to label CV/VC sounds presented in random temporal order. This seems consistent with phone memory dysfunction. (3) These conclusions are at odds with previous studies finding no indication of phone identification impairment.

2pSC4. Preschoolers initiate more conversations than their parents. Mark VanDam, Sarah Campanella, Kiley Wolfenstein (Speech & Hearing Sci., Washington State Univ., P.O. Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Paul De Palma (Comput. Sci, Gonzaga Univ, Spokane, WA), and Daniel Olds (Speech & Hearing Sci., Washington State Univ., Spokane, WA)

Children develop, learn, and refine the complex rules of how to have a conversation beginning in their preschool years. Recent work on conversational exchanges and language usage within families has shown differences (and in some cases, similarities) in how mothers, fathers, girls, and boys interact and converse with each other. It has been suggested that mothers contribute more child-oriented and child-driven exchanges, fathers contribute more formal language that includes problem-solving and linguistic manipulation, girls seek to maintain relationships, and boys seek to establish dominance and attract or maintain and audience. These factors may influence the roles each interlocutor plays in a communicative exchange. One aspect of verbal interaction is who initiates a conversational exchange. This study examined 134 daylong audio recordings using automated speech processing techniques to estimate the frequency of conversation initiation for mothers, fathers, girls, and boys. We found that children initiate conversations most frequently, followed by mothers, followed by fathers. We found no rate difference between girls and boys. Results are consistent with the Bridge Hypothesis or Apprenticeship Model in which interlocutors are motivated in part by their social role, in this case by the social modeling parents demonstrate for their children.

2pSC5. The development of emotional speech prosody perception in 3- to 14-month infants: a preferential listening study. Chieh Kao and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, kaoxx096@umn.edu)

Developmental studies have shown strong evidence that socially enriched speech signal (including prosodic modifications) attracts infants' attention and facilitates language development. While emotion understanding is evident at 9 months of age (Otte *et al*., 2015), the developmental trajectory of the emotional speech prosody perception is still unclear. The present study adopted a widely used preferential looking paradigm to measure 3- to 14-month-old infants' listening preference to English words spoken in neutral, happy, angry, and sad tones. Analysis using a linear mixed model showed that infants' preference of emotional prosody changed as a function of age. On average, the three-month olds listened longer to all emotional prosodies over the neutral one whereas older infants showed significantly diminished interests in the sad prosody, followed by the happy and angry voices. Around 12 months, infants appeared to listen to emotional prosodies equally with the exception of reduced interest in the angry prosody. These preferential listening measures were not correlated with the varying durations or fundamental frequencies of the spoken words for the different emotional categories, indicating that the development of emotional speech prosody is not purely driven by the acoustical properties but rather involves higher-order social cognition.

2pSC6. Syllable production in hard-of-hearing preschoolers. Mark VanDam, Kiley Wolfenstein, Sarah Campanella (Speech & Hearing Sci., Washington State Univ., P.O. Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Daniel Olds (Comput. Sci, Washington State Univ, Spokane, WA), and Paul De Palma (Comput. Sci, Gonzaga Univ., Spokane, WA)

Expressive language in preschoolers has been shown to be positively related to later language development, although not all studies have shown robust effects. Most studies that consider sex have shown that girls develop expressive language earlier than boys. It has also been widely demonstrated that children with a variety of speech-, language-, and hearing-related disorders show deficits in expressive language skills. Children with hearing loss in particular have shown deficits in consonant production, prosodic control, vowel duration, and mean length of utterance. This study examines the expressive language of preschool children with hearing loss using estimates of syllable production from 366 daylong audio recordings totaling over 4348 h of audio processed with unsupervised automatic speech processing techniques. We found no main effect of sex, but typically-developing children were more voluble on average. Unexpectedly, an interaction effect suggests that typically developing boys may be driving the observed differences. This work adds to the evidence of social and biological variability in the speech production of children, and is further proof-of-concept of developmental speech production work using massive data sets and automatic analysis methods.

2pSC7. Child-directed speech enhances preschoolers' speech perception in noise. Nicholas A. Smith (Dept. of Speech, Lang. and Hearing Sci., Univ. of Missouri, Columbia, MO 65211, smithnich@health.missouri.edu), Christine A. Hammans, Timothy J. Vallier (Boys Town National Res. Hospital, Omaha, NE), and Bob McMurray (Dept. of Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

A dominant idea in research on infant-directed speech is that caregivers hyperarticulate their speech and exaggerate speech clarity as a means of facilitating language learning for infants. Much less is known about the acoustic-phonetic properties of speech to young children who are more amenable to perceptual testing. This study tests whether child-directed speech (CDS) is more intelligible by testing 4- to 6-year-old listeners in a speech-in-noise task in which they selected the corresponding picture on a touchscreen. Target words were two-way minimal pairs that contrasted in terms consonant voicing (e.g., "back" versus "pack"), vowel (e.g., "back" versus "beak"), or consonant and vowel (e.g., "back" versus "peak"). At a constant signal-to-noise ratio of -6 dB, speech recognition performance was significantly greater for CDS than for adult-directed speech (ADS). However, significant differences were also found across stimulus talkers, with some mothers providing a greater overall CDS benefit than others. Furthermore, CDS was related to greater enhancement of vowel contrasts in some mothers, and greater enhancement of consonant voicing in others. These perceptual results are discussed in terms of the acoustic-phonetic properties of these speech productions.

2pSC8. Contextual influences on infants' attention to child-directed speech. Robin Panneton, Tyler McFayden, Madeleine Bruce, and Caroline Taylor (Dept. of Psych., Virginia Tech, Blacksburg, VA 24061-0436, panneton@vt.edu)

Initial publications of infants' preference for infant (child)-directed speech (CDS) over adult-directed speech (ADS) produced a swift generalization to all early development. In fact, a recent meta-analysis found that the CDS preference to be robust (Dunst *et al*., 2012) and the ManyBabies I replication effort extends this view across labs, methods, and samples (Frank *et al*., 2018). Nonetheless, there are important demonstrations of how this preference gets attenuated and augmented by various factors. For example, Cooper *et al*. (1997) found that 1-mo-olds did not prefer CDS over ADS when both recordings were of the infants' own mothers, but did prefer CDS when the speakers were unfamiliar women. Other studies on CDS perception find influences of the language being spoken, the modality of presentation, the age of the infant, the clinical status of the mother and/or of the infant. The influence of context on infants' attention to speech is an underappreciated and understudied aspect of early language learning. This presentation will summarize contextual moderators of the CDS preference that emanate from the infant and from the infant's developmental milieu, and offer suggestions for other contexts that most likely impact infants' attention to CDS but have yet to be investigated (e.g., poverty).

2pSC9. Using acoustic features of mothers' infant-directed speech to predict changes in infant biobehavioral state. Jacek Kolacz (Traumatic Stress Res. Consortium at the Kinsey Inst., Indiana Univ., Lindley Hall 428, 150 S Woodlawn Ave., Bloomington, IN 47405, jkolacz@iu.edu), Elizabeth B. daSilva (Indiana University-Purdue Univ. Columbus, Columbus, IN), Gregory F. Lewis (Intelligent Systems Eng., Indiana Univ., Bloomington, IN), Bennett I. Bertenthal (Dept. of Psychol. and Brain Sci., Indiana Univ., Bloomington, IN), and Stephen W. Porges (Traumatic Stress Res. Consortium at the Kinsey Inst., Indiana Univ., Bloomington, IN)

Infant directed speech is marked by exaggerated frequency modulation and strong high frequency power, features that may provide physiological cues for mobilization or calming (Porges and Lewis, 2010; Kolacz *et al*., 2018). We examined whether these features predicted changes in infant

biobehavioral state during the Still Face Paradigm, a stressor in which the mother withdraws and reinstates social cues. 98 mother-infant dyads participated when infants were 4-8 months old. Infant heart rate and respiratory sinus arrhythmia (a measure of cardiac parasympathetic control) were derived from an electrocardiogram (ECG). Infant behavioral distress was measured by vocal, facial, and body movement distress. Mothers' vocalizations were measured using spectral analysis within designated frequency bands and modulation using a 2-dimensional fast Fourier transform of the audio spectrogram. Maternal frequency modulation predicted decreases in infant heart rate ($p = .030$), mid-frequency acoustic power (500–5000 Hz) predicted increases in cardiac parasympathetic regulation in infants with low parasympathetic tone ($p = .024$), and high frequency power predicted increases in infant behavioral distress in infants who were not initially distressed ($p = .011$). These results suggest that mothers' vocal frequency band power and modulation may be aspects of infant-directed speech that are relevant for regulating infant biobehavioral state.

2pSC10. The effects of parent coaching on language outcomes at 18 and 24 months: A randomized controlled trial. Naja Ferjan Ramirez, Sarah Lytle, Ruofan Cai, and Patricia K. Kuhl (Univ. of Washington, Box 357988, Seattle, WA 98115, naja@u.washington.edu)

The prevalence of parentese in speech directed to 11- and 14-month-old infants predicts infants' concurrent babbling as well as their future language skills at 24 months, suggesting that this speaking style may enhance learning. We recently showed that when parents are "coached" about the importance of language input to infants, and parentese, they increase the proportion of parentese and child-directed speech. This has an immediate and positive effect on child language outcomes at 14 months. In the present study, we asked whether the effects of coaching parents extend to longer-term language outcomes. Families of typically developing 6-month-old infants were assigned to Intervention (parent coaching) and Control (no coaching) groups. Parent coaching took place when infants were 6- 10-, 14- and 18-months of age, and included quantitative and qualitative linguistic feedback derived from each family's first-person LENA recordings at home. Language outcomes were measured at 18- and 24-months of age. Parent coaching significantly enhanced the percentage of child-directed speech and parentese in parental input between 6 and 18 months in coached vs. uncoached parents. Children of parents who received coaching showed enhanced language outcomes at 18 and 24 months.

2pSC11. The real-time dynamics of child-directed speech: Using pupillometry to evaluate children's processing of natural pitch contours. Mira L. Nencheva, Elise A. Piazza, and Casey Lew-Williams (Dept. of Psych., Princeton Univ., Princeton, NJ 08540, nencheva@princeton.edu)

Young children prefer child-directed speech (CDS) to adult-directed speech (ADS) (Cooper and Aslin 1990), and its structural and prosodic features are known to facilitate learning (Thiessen *et al.*, 2005; Graf Estes and Hurley, 2013). However, little is known about how the real-time dynamics of CDS prosody affect young children's engagement and learning. In Experiment 1, we evaluated moment-to-moment processing of pitch variation using measures of pupil size synchrony across children (Kang and Wheatley, 2017). 24-to-30-month-old children listened to a story in CDS and ADS, and we found that pupil synchrony was higher for CDS than ADS. Next, using hierarchical clustering, we uncovered 4 main word-level pitch contours from a natural CDS corpus and identified contours (specifically, U- and inverted-U-shaped contours), which elicited lower vs. higher synchrony, respectively. In Experiment 2, we found that children learned novel words better when they were presented in higher- vs. lower-synchrony contours. Importantly, synchrony for a novel word during training significantly predicted learning for the same word at test. By revealing a physiological response that is sensitive to the real-time dynamics of prosody, this investigation yields a new subsecond framework for understanding children's engagement with a signal known to support early language learning.

2pSC12. The effects of parental interaction on French-English bilingual infants' vocalization and turn-taking rates. Katherine Xu, Adrial J. Orena, Yufang Ruan, and Linda Polka (School of Commun. Sci. & Disord., McGill Univ., 2001 Mcgill College Ave., 8th Fl., Montreal, QC H3A 1G1, Canada, tian.y.xu@mail.mcgill.ca)

Prior studies suggest that parent-infant interactions play an important role on infant volubility, which is an important indication of language development. However, little is known about how speaker and language contexts affect infant volubility in bilingual infants. In the current study, we examined how the speaker context (Mother versus Father) and the language context (French versus English) might influence bilingual infants' rate of vocalization and turn-taking. We analyzed naturalistic daylong recordings from English-French bilingual parent/infant dyads ($n = 21$). Preliminary analysis showed that mothers elicited more turn-taking in their 10-month-old infants than fathers did, but they did not elicit more vocalizations. Importantly, there was no effect of language on infant vocalization or turn-taking. We will alsoreport analyses that examine whether the dominant language in the bilingual child's input has an effect on infant volubility that is independent of talker effects. These novel findings will inform our understanding of how parent-infant interactions shape the language development of infants being raised in bilingual families.

2pSC13. Acoustic correlates of hypokinetic articulation of continuously spoken sentences. Amitava Biswas, Mary Schaub, and Steven Cloud (Speech and Hearing Sci., Univ. of Southern Mississippi, 118 College Dr. #5092, SHS Lab., Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Hypokinetic speech involves reduced range of motion of the articulators such as tongue, lips, and jaw. As a result, the formant transitions are relatively reduced. This characteristic may be observed in some adults and children. An algorithm has been developed to estimate the relative variations in formant transitions in continuously spoken sentences. Its sensitivity, specificity, and applications in clinics will be discussed.

2pSC14. Infant-directed speech register in children with and without hearing loss. Maria V. Kondaurova, Kaelin Kinney, Abigail Betts (Psychol. & Brain Sci., Univ. of Louisville, 317 Life Sci. Bld., Louisville, KY 46292, maria.kondaurova@louisville.edu), Lindsay Nolan (Heuser Hearing Inst. & Lang. Acad., Louisville, KY), and Mark VanDam (Elson S. Floyd College of Medicine, Dept. Speech & Hearing Sci., Washington State Univ., Spokane, WA)

Do children with hearing loss use infant-directed speech? The study examined speech characteristics of a 6-year-old child with bilateral cochlear implants and an age-matched child with normal-hearing while interacting with their infant siblings (age 29 and 20 months) and with their mothers. Child-sibling and child-mother interactions were recorded in two conditions. In the "toy" condition, the children explained to their siblings and their mothers how to assemble a toy. In the "book" condition, the children narrated a story using a picture book. Sixty-five vocalizations from each child's speech sample were extracted in each condition. Mean fundamental frequency, fundamental frequency range, utterance duration, number of syllables per utterance, and speech rate were measured. Both children produced higher fundamental frequency, expanded fundamental frequency range, shorter utterance duration, and slower speech rate in the sibling- compared to mother-directed speech in both the "book" and "toy" conditions. For the mother-directed speech only, the children produced lower fundamental frequency, longer utterance duration and more syllables per utterance in the "book" than the "toy" condition. The results suggest that children with and without hearing loss modify prosodic characteristics of their speech when interacting with a younger sibling but the strength of the modification may be task-dependent.

2pSC15. Acoustical regularities in infant-directed vocalizations worldwide. Cody J. Moser (Dept. of Anthropology, Texas A&M Univ., 340 Spence St., College Station, Texas 77840, mose3774@tamu.edu), Harry Lee-Rubin (Dept. of Psych., Harvard Univ., Cambridge, MA), Infant-Directed Vocalizations Collaboration (none, Cambridge, MA), Constance M. Bainbridge, Stephanie Atwood, Max M. Krasnow, and Samuel A. Mehr (Dept. of Psych., Harvard Univ., Cambridge, MA)

Adults often differentiate their song and speech between infants and other adults. Why? Is this a product of just some cultures or does it reflect a universal part of human vocal communication and human cognition? On the latter hypothesis, infant directed-song might regularly exhibit certain features across cultures, including high redundancy and repetition, high signal-to-noise ratios, and superb vowel prolongation and stability compared to adult-directed song. We built a corpus of 1614 recordings of infant- and adult-directed singing and speech produced by 411 people living in 20 societies, including hunter-gatherers and subsistence farmers. Each participant provided examples of each of the four vocalization types. Using exploratory and confirmatory analyses, we show that the acoustical features of infant-directed song and speech are universally distinct from adult-directed song and speech, especially in terms of the phonetic space of their formants, their general rhythmic structure, and their pitch range attributes.

2pSC16. Separability of infant-directed from adult-directed speech is affected by number of channels in cochlear-implant simulated speech. Meisam K. Arjmandi, Laura Dilley (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd. Oyer Speech & Hearing, Rm. 211A, East Lansing, MI 48824-1220, khalilar@msu.edu), Yuanyuan Wang (Ohio State Univ., Columbus, OH), Mario Svirsky (New York Univ., New York, NY), Matt Lehet (Communicative Sci. and Disord., Michigan State Univ., Pittsburgh, Pennsylvania), and Derek Houston (Ohio State Univ., Columbus, OH)

Many studies have demonstrated benefits of infant-directed speech (IDS) over adult-directed speech (ADS) for language development. For deaf children who use cochlear implants (CIs), a variety of factors might reduce the advantage of IDS in language development. We hypothesized that spectral degradation due to the number of channels in CI processing negatively affects acoustic separability of IDS and ADS, potentially reducing benefits for language development of IDS in early childhood. 493 sentences spoken in two speaking styles (IDS and ADS) were processed using 4, 8, 12, 16, 22, and 32 channels noise-vocoders to simulate the spectral degradation caused by CIs. The sentences were partitioned into frames of 30 ms and represented by Mel-frequency cepstral coefficients. The Mahalanobis distance was used to calculate the acoustic distance across speaking styles (ID versus AD) and processing condition (4, 8, 12, 16, 22, and 32 channels). The results show that spectral degradation imposed by CI processing has significant negative effects on separation of IDS from ADS. These findings suggest that the spectral information in speech received by infants with CIs is substantially degraded, which can weaken the facilitative link between IDS and language development in this at-risk population. [Work supported by NIH grant R01DC008581.]

2pSC17. Infant-directed speech enhances recognizability of individual mothers' voices. Thayabaran Kathiresan (Univ. of Zurich, Zurich, Switzerland), Laura Dilley (Michigan State Univ., East Lansing, MI), Simon Townsend (Univ. of Zurich, Zurich, Switzerland), Rushen Shi (Universite du PQ a Montreal, Montreal, QC, Canada), Moritz Daum (Univ. of Zurich, Zurich, Switzerland), Meisam K. Arjmandi (Michigan State Univ., East Lansing, MI), and Volker Dellwo (Phonet. Lab., Univ. of Zurich, Plattenstrasse 54, Zurich 8005, Switzerland, volker.dellwo@uzh.ch)

Adult speakers commonly alter their voices when talking to infants, giving rise to an infant-directed speech (IDS) style. Here we tested the effects of infant-directed speech on the recognizability of a speaker's voice. 10 Swiss-German mothers were recorded talking to their infants IDS and talking to an adult experimenter (in adult-directed speech, ADS). We studied the indexical properties using Mel-frequency cepstral coefficients (MFCCs). By using an unsupervised K-means clustering algorithm, the segmental 13-dimensional MFCCs were clustered and reduced to two dimensions using Principal Component Analysis. Results showed that the relative area of IDS occupied in the 13-dimensional space was significantly larger compared to the area occupied by ADS. A supervised language-independent Gaussian Mixture Model revealed that this expansion benefitted the recognizability of mothers' voices. This means that the higher indexical variability in IDS fosters recognition of individual mothers. Results are consistent with the view that IDS may have evolved in part as a strategy to promote indexical signaling by a mother to her offspring, thereby promoting mother-infant attachment and fostering offspring survival.

2pSC18. Vowel space and variability in infant- and adult-directed speech. Kelly D. Burkinshaw (School of Lang., Linguist, Literatures and Cultures, Univ. of Calgary, Craigie Hall C211, 2500 University Dr. N.W., Calgary, AB T2N 1N4, Canada, kburkins@ucalgary.ca), Lori L. Holt (Dept. of Psych., Carnegie Mellon Univ., Pittsburgh, PA), and Suzanne Curtin (Dept. of Psych., Univ. of Calgary, Calgary, AB, Canada)

Infant- and adult-directed speech are acoustically-distinct registers, but whether the characteristics of infant-directed speech (IDS) promote speech category learning is unclear. Several studies have reported an expanded vowel space in IDS compared to ADS; point vowels (/i/, /a/, /u/) are, on average, more distinct from each other in IDS. But, other studies report greater intra-category variability which might diminish any benefits of an expanded vowel space. Here, we examined vowel productions across five vowels as mothers spoke to their infants (7- or 15-months) in IDS and an adult experimenter in ADS. We observed an expanded point vowel space and an increase in variability of within-category vowel productions in IDS toward 15-month-olds, compared to ADS. Thus, although the centroids of the point vowels were more separated in IDS toward 15-month-olds than ADS, production variability led to substantial category overlap. Yet, classification modeled using Discriminant Function Analysis (DFA) revealed near-ceiling classification rates for both registers, indicating that there was no classification advantage of the increased distance among IDS point vowels. We found neither vowel space expansion nor increased variability in IDS toward 7-month-olds. These findings inform how distributional characteristics of speech input may contribute to vowel category learning in infancy.

2pSC19. Investigating naturalistic code-switching directed towards infants. Lena V. Kremin (Psych., Concordia Univ., 7141 Sherbrooke St. W, PY-033, Montreal, QC H4B 1R6, Canada, lena.kremin@mail.concordia.ca), Adriel John Orena (School of Commun. Sci. and Disord., McGill Univ., Montréal, QC, Canada), Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada), and Krista Byers-Heinlein (Psych., Concordia Univ., Montreal, QC, Canada)

Mixing two languages in speech (i.e., code-switching) is prevalent in multilingual settings, including in speech directed towards infants. Prior research suggests a link between parental code-switching and vocabulary size (Byers-Heinlein, 2013). Moreover, laboratory work suggests that some types of code-switching appear more difficult for infants to process than others (Byers-Heinlein *et al*., 2017; Potter *et al*., 2018). This raises the possibility that the effects of parental code-switching depend on the parents' specific behavior in terms of the frequency, location, and purpose of code-switching (Byers-Heinlein, 2017). Prior studies of parental code-switching relied on self-report or short lab observations. In this study, we analyze parental code-switching behavior in a corpus of daylong home recordings of 21 infants (at 10- and 18-months) from French-English bilingual families in Montréal. We will identify instances of parental code-switching, their syntactic location, the direction of the switch, and the apparent reason for the switch (e.g., teaching vocabulary, translating an entire utterance). Preliminary results indicate that the frequency of code-switching varies between families and that code-switching between sentences is more common than code-switching within a sentence. This project will provide the first in-depth investigation about the characteristics of naturally produced parental code-switching.

2pSC20. Average daily speech exposure for fetuses. Brian B. Monson and Molly Cull (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61822, monson@illinois.edu)

The quality and quantity of speech and language exposure during early childhood is believed to be predictive of language ability during later childhood development. However, the human auditory nervous system comes "online" *in utero,* at least as early as 23 weeks' gestation. It has been demonstrated that intrauterine fetal experience with extrauterine sounds during this last trimester of gestation is sufficient to impact auditory brain development and neural responses to speech. Whether fetal exposure to speech affects later childhood language development remains an open and difficult question. To begin to address this question, we collected and analyzed fetal auditory exposure data for sounds generated in the extrauterine environment using small audio recorders worn by pregnant women during the third trimester of pregnancy. Averaged across 2200 h of audio data, daily speech exposure for individual fetus subjects ranged from 2.8 to 5 h, suggesting some newborns may begin extrauterine life with less than 60% of the speech and language exposure of their peers. Whether this variability is associated with variability in subsequent development remains to be seen.

2pSC21. Context-dependent hyperarticulation of the Korean three-way laryngeal stop contrast in clear speech. Seulgi Shin and Allard Jongman (Linguist, Univ. of Kansas, 1846 Tennessee St., Apt. #4, Lawrence, KS 66044, seulgi.shin@ku.edu)

We investigate whether and how speakers' adaptation to feedback is reflected in hyperarticulation by comparing the productions of Korean aspirated, lenis, and fortis stops in plain versus clear speech. Ten Seoul Korean speakers were asked to read a stop-initial nonword and repeat the nonword after receiving misrecognition feedback that contained a contrasting stop. f0 and VOT were examined as main cues to the Korean stop distinction. Results showed that VOT was lengthened for aspirated and lenis stops in clear compared to plain speech. This difference in VOT was attributable to speakers' modification in response to different types of feedback. VOT for aspirated stops was lengthened after both fortis and lenis feedback whereas VOT for lenis stops was lengthened only after fortis feedback. f0 did not show a difference between plain and clear speech. However, f0 enhancement for lenis and fortis stops depended on the type of feedback. f0 for lenis stops was lowered after aspirated feedback and raised after fortis feedback. Fortis stops' f0 was lowered after aspirated and raised after lenis feedback. Overall, speakers can make local modifications in VOT and f0 in response to specific feedback, suggesting precise control in their effort to distinguish sounds and maintain intelligibility.

2pSC22. Characterising maternal pitch contours used during interactions with infants at high and low risk for autism spectrum disorder. Alix Woolard (Psych., Univ. of Newcastle, University Dr., Callaghan, NSW 2308, Australia, alix.woolard@uon.edu.au), Alison Lane (Health Sci., Univ. of Newcastle, Callaghan, NSW, Australia), Linda Campbell, Frini Karayanidis (Psych., Univ. of Newcastle, Callaghan, NSW, Australia), Daniel Barker (Medicine and Public Health, Univ. of Newcastle, New Lambton, NSW, Australia), Larissa Korostenski (Neonataology, John Hunter Children's Hospital, New Lambton, NSW, Australia), Shelly Lane (Health Sci., Univ. of Newcastle, Callaghan, NSW, Australia), and Titia Benders (Linguist, Macquarie Univ., North Ryde, NSW, Australia)

Infant-directed speech (IDS) is the speech register used when interacting with infants. Pitch contours are a salient aspect of IDS and facilitate infant language and socio-communicative development. Little research investigates pitch contours within the context of socio-communication or language deficits, such as infants at high-risk (HR) for Autism Spectrum Disorder (autism). The aim of this study was to characterise pitch contours used by mothers when interacting with HR infants compared to mothers interacting with low-risk (LR) infants. 18 mothers and their 12-month-old infant (12m, 6f) participated in 15-minute recorded interactions. Autism risk was assessed via parent and observer-report. Pearson product-moment correlations were performed to determine relationships between maternal pitch contours and autism risk. Increased risk for autism was associated with fewer utterances (r = -0.576, N = 18, p = 0.01) and fewer rising (r = -0.586, N = 18, p = 0.01), sinusoidal (r = -0.636, N = 18, p = 0.005), and flat contours (r = -0.679, N = 18, p = 0.01), and more complex (r = 0.584, N = 18, p = 0.01), rapid (r = 0.526, N = 18, p = 0.03), and u-shaped contours (r = 0.619, N = 18, p = 0.02). These preliminary data suggest that mothers of HR infants use different patterns of pitch contours than LR mothers. Further assessment of IDS used with HR infants is warranted to identify at what stage IDS patterns deviate between groups.

Session 2pSP

Signal Processing in Acoustics, Acoustical Oceanography, Physical Acoustics, Underwater Acoustics, and Structural Acoustics and Vibration: Borehole Acoustics Logging for Hydrocarbons Reservoir Characterization

Said Assous, Cochair
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Chair's Introduction—1:30

Invited Papers

1:35

2pSP1. Logging while drilling shear sonic logging in very large boreholes. Matthew Blyth (Schlumberger, 32702 Whitehaven Pl., Fulshear, TX 77441, mblyth@slb.com), Naoki Sakiyama (Schlumberger, Sagamihara, Japan), Hiroaki Yamamoto (Schlumberger, Houston, TX), Atsushi Oshima (Schlumberger, Sagamihara, Kanagawa, Japan), and Eduardo Saenz (Schlumberger, Houston, TX)

As LWD sonic tools have become more advanced, they have become more routine in use and are now commonly used in multiple hole sections, with the results having applications across a wide range of problems in well construction. The acquisition of quadrupole shear measurements in large boreholes and slow formations is highly challenging however, due to the borehole conditions, the very slow formation shear values and the effects of the drilling fluid. This paper will discuss the limits of the measurement under these conditions, using both modeled data and real well examples and the effect on measurement quality as a result of these conditions will be shown. The modeling studies shown indicate the importance of understanding the drilling mud properties (slowness and density) was well as the formation and borehole properties when attempting to derive robust shear information under these conditions. The results show that, although the acquisition of reliable shear data in these conditions is challenging, it is not impossible, provided that the LWD tool is correctly designed, the physics are understood and suitable processing applied.

1:55

2pSP2. Simulations of collar waves for acoustic logging while drilling in the frequency and the spatial domains. Xiao He, Yunjia Ji, Hao Chen, and Xiuming Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., 21, Northern 4th Ring Rd. West, Haidian District, Beijing 100190, China, hex@mail.ioa.ac.cn)

Tool waves generated by the presence of the drill collar can cover the formation signals we need. The suppression for those collar waves has been a challenge for designing acoustic logging while drilling (ALWD) tools. To study the collar wave characteristics, we evaluate the excitation intensity of the modes with varying frequencies and positions by solving the elastic wave equations. Analysis results show that the collar wave energy concentrates in the cylindrical tool model. And the radial location of its peak gradually moves from the inner surface to the outer wall with the increasing frequency. The particle vibration trajectories contributed by the collar wave are also investigated. It is revealed that the collar mode in the lower frequency range has an approximately longitudinal polarization, while it tends to the transverse-wave-like motion at high frequencies. According to these features, we propose strategies on the tool structure designs to suppress the collar waves in the lower- and the higher- frequency ranges, respectively. Numerical examples further validate that the proposed collar structures with interlayers or grooves can well attenuate the collar wave propagation.

2:15

2pSP3. Determining fracture properties from acoustic borehole waves: theory and experimental results. Huajun Fan (College of Geophys. and Information Eng., China Univ. of Petroleum (Beijing), 18 Fuxue Rd., Changping, Beijing 102249, China, hj.fan@hotmail.com) and David Smeulders (Dept. of Mech. Eng., Eindhoven Univ. of Technol., Eindhoven, The Netherlands)

A borehole in the subsurface may penetrate rocks which are porous, permeable and fractured. Pressure transients in the borehole will therefore cause viscous fluid to flow into and out of the wall of the borehole. This forced flow consumes some energy and affects the phase velocity and amplitude of the waves traveling in the borehole fluid column. These effects can be evaluated to extract information about the rocks adjacent to the borehole, which is of paramount importance for the oil and gas industry. This work discusses laboratory wave experiments in a 7.5 m long vertical shock tube. Rock samples having a vertical borehole and horizontal fractures are installed in the shock tube and filled with water. The shock tube generates broadband pressure transients in the borehole. These are measured in the borehole at variable depth by means of a sliding pressure probe. Repetitive experiments are combined into borehole microseismograms.

In this way borehole wave reflection and transmission coefficients over the fractures are determined. The results show good agreement with low-frequency theory, where it is assumed that the wavelengths are much larger than the tube diameter. Inversely, it is shown that the aperture and the length of the fracture can directly be inferred from acoustic borehole experiments.

2:35

2pSP4. Features of the protective cover of the ultrasonic imager with high resolution. Kamil Yusupov, Victor Kosarev, and Adel Akchurin (Kazan Federal Univ., 18 Kremlyovskaya St., Kazan 420008, Russian Federation, kamil.usupov@kpfu.ru)

This work describes the features and design of the acoustically semitransparent cover of a borehole ultrasonic imager with high resolution. This scanner is designed to investigate the fine structure of the well surface by ultrasonic sounding method at a frequency of 800 kHz. The ultrasonic transducer is rotating around a central axis. However, in the design of the device fixed semitransparent protective cover is provided for prevent damage of mechanical parts. Nevertheless, during reflected signals recording from the wall of well the undesirable noise reflections appear from protective cover. These noises interfere with the algorithm of detecting the main signal. The logging tool electronics is based on a single FPGA chip, its logic cells capacity allows determining the maximum reflection amplitude and its location, as well as recording the signal time window to the SD-card (with 32GB storage space). This windows are processed at the personal computer after logging. Such approach allows analyzing waveforms and noises. After a multiple laboratory and well experiments this logging tool was allow improving the design of the acoustically semitransparent protective cover for maximum attenuation of the reflection noise. In addition, the forms of the recorded acoustic signals and the design of the protective cover are given.

2:55

2pSP5. Assessing organic richness of source rocks through integration of acoustic logs and microresistivity images. M. S. McQuown (Weatherford, 10844 Diane Dr., Golden, CO 80403, scott.mcquown@weatherford.com)

Accurate evaluation of source rock thickness and organic richness are key components to estimating the hydrocarbon potential of a basin and identifying target zones in self-sourcing shale plays. The millimeter-scale variability in mineralogy and organic matter concentrations in source rocks can make petrophysical quantification problematic when relying on the resolution of standard well logs. Integrating acoustic well log data with microresistivity images with can improve resolution while generating more meaningful information than their separate results. We developed a new algorithm that utilizes the superior resolution and circumferential coverage of microresistivity images and integrates them with enhanced acoustic data to assess the organic richness of source rocks. The result is an oriented 360 deg pseudo-sonic image that reveals anomalous properties related to organic richness. The validity of the method is assessed by comparing the results with standard well log-derived solutions for total organic carbon. Excellent correlation between the image results and petrophysical solutions is observed.

Contributed Papers

3:15

2pSP6. A method to locate acoustic emission events induced by hydraulic fracturing using waveform correlation weighted by amplitude stacking. Chengwei Zhang, Wenxiao Qiao, and Xiaohua Chen (College of Geophys. and Information Eng., China Univ. of Petroleum-Beijing, No. 18, Fuxue Rd., Changping District, Beijing 102249, China, zhangchengwei.cq@163.com)

Acoustic emission monitoring has been the best technique to evaluate hydraulic fracturing which is the most common method used to induce fractures for stimulating hydrocarbon reservoirs. Correlation-based and amplitude-stacking-based methods, both are based on migration and are all suitable for locating acoustic emission events induced by hydraulic fracturing without the need to pick the arrival times of the P- and S-waves. By comparing and analyzing the advantages and disadvantages of the two location methods, we propose a new weighted correlation method using both amplitude and waveform correlation. First, we calculate the travel time of acoustic waves from the trial point in the formation to each receiver by the ray-tracing method, and further to determine the time-window positions of the P- and S-waves on all waveforms. Then, we calculate the correlation of the waveforms in the windows and the amplitude stacking of the average energy ratio between the short-time window and the long-time window on the original acoustic waveforms. Finally, we use the correlation weighted by amplitude stacking to image space locations of acoustic emissions. Tests with synthetic data show that the weighted correlation method has stronger stability and lower location uncertainty than the existing migration-based location methods.

3:30

2pSP7. Borehole sonic array processing and the group versus phase velocity debate. Said Assous (GeoSci., Weatherford, East Leake, Loughborough LE126JX, United Kingdom, said.assous@eu.weatherford.com), Laurie Linnett (none, Scotland, United Kingdom), and Peter Elkington (GeoSci., Weatherford, Loughborough, United Kingdom)

We consider the impact of processing method on the ability to differentiate between group and phase velocity in dipole sonic log data. A new array processing method is used to demonstrate that the choice of processing algorithm could lead to confusion. We emphasise the importance of knowing the limitations of the processing method used before interpreting results and applying them to the calculation of elastic rock properties. The proposed method addresses the confusion and clarifies that what is computed from the array waveforms can be either group or phase velocity depending on the algorithm used and the formation properties in the given situation. We demonstrate the effectiveness of the approach versus conventional processing methods on synthetic examples.

3:45–4:00 Panel Discussion

Session 2pUW

Underwater Acoustics: Reflection and Scattering from Ocean Surface and Bottom

Derek R. Olson, Chair
Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943

Contributed Papers

1:30

2pUW1. Estimates of coherent reflection loss inclusive of the effects of near-surface bubbles on sound speed. Adrian D. Jones and Alex Zinoviev (Maritime Div., Defence Sci. and Technol. Group, P.O.Box 1500, Edinburgh, SA 5111, Australia, bearjones@adam.com.au)

Through simplification of the authors' model "JBZ" of coherent reflection loss at the ocean surface, a revised model has been obtained in extremely simple form. This model provides descriptions of the loss obtained at a wind-roughened surface, adhering to a Pierson-Moskowitz surface wave frequency spectrum, inclusive of the effects of near-surface bubbles on sound speed. A very compact form of the model has been obtained for application to surface ducted sound for a wide range of wind speed and frequency combinations, with the result that the dB loss with range is the same for all modes within the duct. The derivation of the model from the authors' earlier JBZ is described, and the conditions for equivalence are detailed. In a limited range of comparisons, descriptions of reflection loss per bounce obtained with the original JBZ and the compact version are shown to be nearly identical to corresponding values obtained by the "Relay" model described by Ainslie [*JASA* **118**, 3513–3523 (2005)].

1:45

2pUW2. Numerical investigation of the two-scale model for rough surface scattering. Derek R. Olson (Oceanogr., Naval Postgrad. School, 201 Appl. Sci. Bldg., University Park, Pennsylvania 16802, olson.derek.r@gmail.com)

The two-scale approximation for rough surface scattering assumes that a rough surface can be divided into two components: a large-scale and small-scale component, using a cutoff wavenumber to partition the wavenumber domain. This approach is attractive for multi-scale roughness such as the air-ocean and ocean-seafloor interfaces. Traditionally, the small roughness perturbation approximation is used for the small-scale component, and the Kirchhoff, or tangent-plane approximation is used for the large-scale component. Typically, the high-frequency approximation is used when propagating the surface fields to the far field, resulting in the physical interpretation of the large-scale surface modulating the local grazing angle of the small-scale surface. Using these approximations, the incoherent scattered power exhibits a dependence on the cutoff frequency, which is undesirable from a theoretical point of view. In this work, the validity of the approximations used on the small-scale surface, and large scale surface is examined through the use of numerical solution of the governing integral equations. This investigation is performed with the goal of testing the hypothesis that if each approximation on the component surfaces is demonstrated to be accurate, then the two-scale model should be independent of the cutoff frequency.

2:00

2pUW3. Time warping and acoustic characterization of the seafloor in horizontally inhomogeneous ocean. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu), Boris Katsnelson (Univ. of Haifa, Haifa, Israel), Tsu Wei Tan (Phys. Dept., Naval Postgrad. School, Monterey, CA), and Andrey Malykhin (Phys. Dept., Voronezh State Univ., Voronezh, Russian Federation)

Time-warping transform is increasingly employed in shallow water acoustics to separate the field due to a compact, broadband sound source or the two-point cross-correlation function of diffuse noise into their normal mode components, and to measure mode travel times as a function of frequency. The time-warping transform was developed for range-independent waveguides, while physical parameters of the ocean are never quite constant in the horizontal plane. Bathymetry variations are typically responsible for the bulk of the waveguide's range dependence as well as horizontal refraction of sound in the coastal ocean. Simple, exactly solvable models of shallow-water waveguides are used in this paper to illustrate the effects that the range dependence and horizontal refraction have on the performance of the warping transform and on the inferred geoacoustic parameters. Horizontal refraction due to generic bathymetric variations is addressed in the adiabatic approximation using perturbation techniques. Theoretical predictions are verified using numerical simulations. It is found that moderate bottom slopes can lead to large errors in retrieved geoacoustic parameters and cause positive bias in bottom sound speed estimates if horizontal refraction is ignored. [Work supported, in part, by NSF and BSF.]

2:15

2pUW4. Measurements of two-dimensional spatial coherence of normal-incidence seafloor scattering. Daniel C. Brown, Cale F. Brownstead (Penn State Univ., State College, PA 16804, dcb19@psu.edu), Anthony P. Lyons (Univ. of New Hampshire, Durham, NH), and Thomas B. Gabrielson (Penn State Univ., State College, PA)

The two-dimensional spatial coherence of the field backscattered from a complex lakebed has been characterized in a series of measurements made in Seneca Lake, New York. In the test region, the lakebed consists of a series of sediment layers created by a sequence of distinct depositional processes. The spatial coherence depends on the structure of the underlying sediment sequences. Significant ping-to-ping variability in the spatial coherence surface is observed for each sediment sequence. This variability is quantified by a two-dimensional spatial coherence metric that measures the coherence lengths and asymmetric coherence surface orientation. Sediment sequences with isotropic scattering strength exhibit random ping-to-ping variability in coherence length and coherence surface orientation. Sequences with spatially anisotropic scattering strength show intervals of non-random ping-to-ping variability in the coherence length and coherence surface orientation.

2:30

2pUW5. Sequential bottom parameter estimation using blind deconvolution of sources of opportunity in ocean waveguide for Santa Barbara channel experiment. Xuedong Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, zxd@mail.ioa.ac.cn), Nicholas C. Durofchalk, Karim G. Sabra (Georgian Inst. of Technol., Mech. Eng., Atlanta, GA), and Lixin Wu (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

This paper investigates the performance of sequential bottom parameter estimation based on ray-based blind deconvolution (RBD) [Sabra *et al*., *JASA* EL42-7 (2010)] of sources of opportunity using the 2016 Santa Barbara Channel (SBC) experimental recordings of shipping noise. The RBD algorithm relies on estimating the unknown phase of the source of opportunity through wideband beamforming along a well-resolved ray path to approximate the environment's channel impulse responses (CIR) between the source and the VLA elements. The corrected power ratio of the direct and bottom-bounced arrivals is processed to infer the bottom reflection loss and is utilized to invert for the bottom parameters. Sequential parameter estimation uses a state space model for predicting and correcting the bottom parameters as the estimated bottom reflection loss values become available. Inversions results for the SBC experiment were also performed with conventional active sources to validate the inversion obtained with RBD of sources of opportunity.

2:45

2pUW6. Simulation and testing results for a sub-bottom imaging sonar. Daniel C. Brown, Shawn Johnson (Penn State Univ., State College, PA 16804, dcb19@psu.edu), Isaac Gerg (Penn State Univ., Port Matilda, PA), and Cale F. Brownstead (Penn State Univ., State College, PA)

The problem of detecting buried unexploded ordnance is addressed with a sensor deployed from a shallow-draft surface vessel. The sonar system produces three-dimensional synthetic aperture sonar imagery of both surficial and buried ordnance across a range of environments. The sensor's hardware design is based in part upon data created using a hybrid modeling approach that combined results from separate environmental scattering and target scattering models. This hybrid model produces synthetic sensor data where the sensor/environment/target space may be modified to explore the expected operating conditions. Based on these modeling results, a sonar system has been integrated to a test platform, and experiments have been conducted at a trial site in the Foster Joseph Sayers Reservoir near Howard, PA. Modeling and experimental results will be presented and discussed.

3:00

2pUW7. Modeling variability in spatial coherence measurements for seafloor scattering. Thomas E. Blanford, Daniel C. Brown, and Richard J. Meyer (Appl. Res. Lab., The Penn State Univ., The Penn State Univ., State College, PA 16804, teb217@psu.edu)

Correlation sonar systems, such as correlation velocity logs and synthetic aperture sonar micronavigation, can estimate platform motion using measurements of the spatial coherence of seafloor scattering. During the course of operation, the spatial coherence measurements made by these systems will fluctuate about their average values. A variety of factors, including noise sources, changing environmental properties, and statistical estimation error can all contribute to the variability in these measurements. This presentation will introduce models that can be used to predict the variability in measurements of spatial coherence. These models will be compared with field measurements of spatial coherence collected at normal incidence to the bottom of Seneca Lake. Finally, the application of these models will be discussed as they relate to the design of arrays and signal processing algorithms for correlation sonar systems. (The authors want to acknowledge the financial support for this work by Lockheed Martin Rotary and Mission Systems.)

3:15

2pUW8. Linear and half order fractional viscoelastic equivalents of the extended Biot model. Sri Nivas Chandrasekaran and Sverre Holm (Dept. of Informatics, Univ. of Oslo, Gaustadalléen 23B, Oslo 0373, Norway, srinc@ifi.uio.no)

The extended Biot poro-viscoelastic model [Chotiros and Isakson, *JASA* (2014)] like the standard Biot poroelastic model is biphasic and predicts two compressional waves (fast and slow) and one shear wave. Taking squirt flow into account, the model introduces two additional relaxation modes apart from the Biot crossover relaxation mode (global flow). The frequency dependent relaxation processes and a high number of independent parameters make the finite element implementation and the inversion of parameters for reconstruction a challenge. To overcome this challenge, we identify single phase viscoelastic equivalents for all three wave solutions of the extended Biot model from its dispersion relations. Considering squirt flow and high-frequency turbulent flow, the fast compressional wave and the shear wave solutions at low frequencies are equivalent to a non-standard solid model with eight parameters (four springs and four dashpots) and six parameters (Three springs and three dashpots), respectively. At high frequencies, the Kelvin-Voigt model used for tangential contact stiffness (shear relaxation) leads to non-physical equivalents. Neglecting this shear relaxation mode, the fast compressional wave and the shear wave solutions at high frequencies are equivalent to a fractional half-order solid model of order three and two, respectively. Alternative models for tangential contact stiffness are discussed.

3:30

2pUW9. Experimental study on measurement of transient sound in a reverberation tank. Rui Tang and Xinyue Yu (Harbin Eng. Univ., Nantong Str. Nangang Dist No.145, Harbin 150001, China, tangrui@hrbeu.edu.cn)

As a new application of reverberation tanks, the experimental study on measuring the sound energy level of transient sound sources in a reverberation tank was proposed in this paper. According to the law of energy conservation, the transient sound energy obtained by sound intensity integration in time domain, which is the same with sound energy density spectrum integration in frequency domain. By utilizing the invariant sound field correction factors, the transient sound source level could be obtained in a reverberation tank, and both the reverberant measurement methods in frequency domain and that in time domain were established. The impulsive sound and the spark sound experiments were carried out in UATL (Underwater Acoustics Technology Lab, Harbin Engineering University, China) to confirm the validity of the proposed method. Experiments results showed that the sound source level of the transient sounds could be precisely measured in reverberation tanks, and compared to free field measurement, the deviation was less than 1 dB.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

W. J. Murphy, Chair ASC S3
National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Cincinnati OH 45226

T. Ricketts, Vice Chair ASC S3
Vanderbilt University, 1215 21st Ave. South, Rm. 8310, Nashville TN 37232

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D. S. Houser, Vice Chair ASC S3/SC 1
National Marine Mammal Foundation, 2240 Shelter Island Dr., Suite 200, San Diego, CA 92106

K. Fristrup, Vice Chair ASC S3/SC 1
National Park Service, Natural Sounds Program, 1201 Oakridge Dr., Suite 100, Fort Collins, CO 80525

Accredited Standards Committee S3/SC 1 on Animal Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

Meeting of Accredited Standards Committee (ASC) S12 Noise

S. J. Lind, Vice Chair, ASC S12
The Trane Co., 3600 Pammel Creek Rd., Bldg. 12-1, La Crosse, WI 54601 7599

D. F. Winker, Vice Chair, ASC S12
ETS-Lindgren Acoustic Systems, 1301 Arrow Point Dr., Cedar Park, TX 78613

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

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1, Noise, and ISO/TC 43/SC 3, Underwater acoustics, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 14 May 2019.

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

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will meet starting at 4:30 p.m.

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

Committees meeting on Tuesday are as follows:

Engineering Acoustics (4:30 p.m.)	McCreary
Signal Processing in Acoustics (4:30 p.m.)	Beckham
Acoustical Oceanography	McCreary
Animal Bioacoustics	Clements
Architectural Acoustics	French
Musical Acoustics	Breathitt
Physical Acoustics	Jones
Psychological and Physiological Acoustics	Carroll Ford
Structural Acoustics and Vibration	Stopher

Committees meeting on Wednesday

Biomedical Acoustics	Nunn

Committees meeting on Thursday

Computational Acoustics (4:30 p.m.)	Clements
Noise	Segell
Speech Communication	Carroll Ford
Underwater Acoustics	McCreary

Session 3aABa

Animal Bioacoustics and Signal Processing in Acoustics: Bioinspiration and Biomimetics in Acoustics III

Rolf Müller, Chair
Mechanical Engineering, ICTAS Life Sciences District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061

Chair's Introduction—8:00

Invited Papers

8:05

3aABa1. Biomimetic sonar echo parameters form cognitive maps. Roman B. Kuc (Elec. Eng., Yale, 15 Prospect St., 511 Becton, New Haven, CT 06511, roman.kuc@yale.edu)

A biomimetic audible sonar probes 2.5D targets and processes binaural echoes to extract values of eight parameters to generate two-dimensional cognitive maps. Targets are configured using posts connected by tangential planes. Being tuned to recognize posts and planes, the sonar produces a cognitive map that is composed of these two components. A platform with translational and rotational degrees of freedom employs right-ear dominance to implement a landmark-centric scanning trajectory whose step size adaptively changes with echo information. The sonar tracks the target by maintaining a constant first echo arrival time and equalizes binaural echo times to form singular echoes. When observed, singular echoes identify landmarks defined by post radii and locations. The mapping process employs five states from detection to termination that passes through the singular echo state. Separate states detect post pairs that exhibit echo interference and planes that exhibit echo amplitude differences. The scanning process terminates when the current landmark parameters match those of the first landmark. Two targets configured with three posts and an added plane illustrate the procedure.

8:25

3aABa2. The acoustic world of odontocete biosonar and technical sonar. Michael J. Roan, Rolf Müller, and Hyeon Lee (Mech. Eng., Virginia Tech, 111 Randolph Hall, 460 Old Turner St., Blacksburg, VA 24061, mroan@vt.edu)

Bats, toothed whales (odontocetes), and man-made sonar all use the same acoustical principles of echo-location. The fundamental components of these systems include transmitter frequency response, tailored waveforms, properties of a propagation medium, clutter, background noise, target strength, and receiver properties. Across all of these elements, there are very large differences (i.e., underwater versus in-air propagation). This gives rise to several interesting questions about the performance of these sonar systems. For example, given the bandwidth and duration of transmitted signals for bats and odontocetes, what are the relative target strengths/maximum imparted Doppler shifts for common prey types? How does the ambiguity function of the echo compare to these target sizes and relative Doppler shifts? What effect does the medium has and how do these systems adapt to clutter, interferers, and noise? This talk will focus on the underwater aspects of these systems which include (i) the properties of the underwater propagation medium, (ii) the geometry and material of the boundaries that limit the underwater propagation channel (this includes targets of interest and clutter), and (iii) the time-frequency and spatial properties of the underwater sources.

8:45

3aABa3. Acoustic reflectivity of a harbor porpoise *Phocoena phocoena.* Whitlow Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96744, wau@hawaii.edu), Ronald Kastelein, and Lean Helder-Hoek (SEAMARCO, Inc., Hardewick, The Netherlands)

Acoustic backscatter measurements were conducted on a stationary harbor porpoise (*Phocoena phocoena*) under controlled conditions. The measurements were made with the porpoise in the broadside aspect using three different types of signals as follows: (1) a 475-ms linear frequency-modulated (FM) pulse with a frequency range from 23 to 160 kHz; (2) a simulated bottlenose dolphin (*Tursiops truncates*) click with a peak frequency of 120 kHz; and (3) a simulated killer whale (*Orcinus orca*) click with a peak frequency of 60 kHz. The measurement with the FM pulse indicated that the mean target strength at the broadside aspect decreased from -26 dB to -50 dB as the frequency increased from 23 to 120 kHz in a nearly linear fashion (on a logarithm plot) and dropped rapidly by 27 dB to -77 dB at 150 kHz. Target strength variation with the frequency was similar to that in a previous backscatter measurement performed on a bottlenose dolphin over a similar frequency range (23–80 kHz). The target strength of the smaller harbor porpoise was about 15–16 dB lower than that of the bottlenose dolphin. The difference in the lung volume of the two species when expressed in dB was also approximately 15 dB. The results suggest that the dolphin bubbrt had broadband anechoic properties.

9:05

3aABa4. Linear and non-linear features of teleost acoustic communication and implications in software defined communication protocols. Cameron A. Matthews (Panama City Div., Naval Surface Warfare Ctr., 110 Vernon Ave., Panama City, FL 32407, cameron.matthews@navy.mil)

Many teleost species communicate via acoustic paths. Many times these paths are chosen through evolution in a way that scientists find contrary to practical physics associated with the ocean environments they are born from. This leads to a practical question of why animals would not naturally become the best basis for acoustic communication, particularly in the emerging market of software defined radios and their application in undersea acoustics. We propose a grouping of three distinct acoustic communication "styles" in terms of linearity—periodic linear, periodic non-linear, and aperiodic non-linear—and what the implications are for a dynamic and relatively wide band software defined communication system are.

9:25

3aABa5. Developing a biomimetic acoustic deterrent to reduce bat mortalities at wind turbines. Michael Smotherman (Biology, Texas A&M Univ., 3258 TAMU, College Station, TX 77843-3258, smotherman@tamu.edu), Paul Sievert, Zara Dowling (Dept. of Environ. Conservation, Univ. of Massachusetts, Amherst, MA), Dan Carlson, and Yahya Modarres-Sadeghi (Mech. Eng., Univ. of Massachusetts, Amherst, MA)

High bat mortalities at wind turbines have emerged as an unexpectedly severe environmental impact of developing wind farms all over the world, motivating an urgent need to develop effect deterrent strategies that can be implemented efficiently on a large scale. Echolocating bats use ultrasonic sonar to navigate, and some studies have shown that broadband ultrasonic noise can have a deterrent effect under certain conditions. Thus, ultrasonic noise may be an attractive approach, but attempts to incorporate electronic sound generating solutions have generally failed because of insufficient bandwidth and intensity as well as their sensitivity to the harsh environmental conditions. To get around these constraints, we designed a biomimetic whistle loosely modeled after the bat larynx that could be mounted on the moving turbine blades to passively generate ultrasonic sounds tuned to the acoustic parameters of bat's auditory system. The whistle produces multi-harmonic tones detectable by the most impacted bat species from distances approaching 100 me away. To test whether the whistle actually deters bats or alters their flight paths, we conducted a series of playback studies in the lab and field using microphone arrays and videography. This presentation will focus on the results of these behavioral studies assessing whether or not bats change their flight trajectories in response to hearing the acoustic stimulus produced by the whistle.

Contributed Paper

9:45

3aABa6. Computer simulation of energy efficient speech production during locomotion over water without regular swimming strokes. Amitava Biswas (SHS Lab., Speech and Hearing Sci., Univ. of Southern Mississippi, 118 College Dr. #5092, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Many aquatic animals need to live extended periods of time over water without much concern for fatigue. In contrast, humans are usually prone to fatigue during locomotion over water and unlikely to sustain energy efficient production of speech simultaneously. According to computer simulations, a strategy of backfloating and articulating all four or some of the limbs like oars can move the body efficiently, and the individual is less likely to fatigue and more able to maintain uninterrupted normal speech conversation for a greater period of time. The motion can be easily directed towards the feet to avoid striking the head against an obstacle when distracted by the speech conversation.

10:00–10:30 Panel Discussion

Session 3aABb

Animal Bioacoustics: Animal Bioacoustics Poster Session

Joseph Sutlive, Cochair
Translational Biology, Medicine, and Health, Virginia Polytechnic Institute and State University, 3719 Parliament Rd., Apt. 22, Roanoke, VA 24014

Ruihao Wang, Cochair
VT, 112 Hearthstone Dr, Apt. 210, Blacksburg, VA 24060

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

Contributed Papers

3aABb1. Investigating Atlantic bottlenose dolphin mismatch negativity response to pure tone stimuli. Brittany A. Romanchek and Peter M. Scheifele (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., P.O. Box 670379, Cincinnati, OH 45267, huttonba@mail.uc.edu)

Marine mammals rely heavily on acoustic cues to survive but are susceptible to hearing impairment due to anthropogenic noise. The quality of life and well-being of marine mammals under professional care is based on enrichment through training and communication conspecifics and zoo staff. Mismatch Negativity (MMN) is an electrophysiological exam that records synchronized firing of neurons from the auditory cortex and frontal lobe, allowing for assessment of echoic memory and sound differentiation capability. The objective of this study is to conduct MMN exams, with acoustic stimuli, on the Atlantic Bottlenose Dolphin (*Tursiops truncatus Montagu*) to study echoic memory and develop auditory cognition assessments for marine mammals in captivity. Three male Atlantic Bottlenose dolphins were tested in water at the Indianapolis Zoo. The oddball paradigm in this study included a frequency difference between two pure-tone stimuli (standard 2000 Hz, deviant 500 Hz). The results suggest that a similar MMN waveform morphology and latency (between 100 and 300 ms) are seen in dolphins when compared to humans. Average P1, N1, P2, and MMN values were determined for each dolphin. Developing MMN tests for animals under professional care will allow for application to marine mammals in natural habitats, which are exposed to high levels of anthropogenic noise.

3aABb2. What is a scream? Acoustic characteristics of a human call type. Jay W. Schwartz, Jonathan W. Engelberg, and Harold Gouzoules (Dept. of Psych., Emory Univ., 36 Eagle Row, Atlanta, GA 30322, jwschwa@emory.edu)

Recent research suggests that human screams comprise an innate call type, yet the defining acoustic structure of screams has not been determined. In this study, participants listened to 75 human vocal sounds, representing both a broad acoustical range and array of emotional contexts, and were asked to classify each as either a scream or a non-scream. Participants showed substantial agreement (Fleiss' kappa test, $\kappa = 0.62$) on which sounds were considered screams. To characterize the acoustic structure of screams, we measured the stimuli on 28 acoustic parameters. PCA and GLMM analyses indicated that the classification as a scream was positively correlated with 3 acoustic factors: one corresponding to high pitch and roughness, another corresponding to a wide fundamental frequency range and narrow bandwidth, and a third negatively correlated with roughness. Thus, the role of roughness in identifying screams appears to depend in part on pitch. Twenty-six stimuli were agreed upon by $>$ 90% of participants to be screams, but others evoked mixed responses, suggesting that while screams might represent an innate call type, their acoustic structure is not narrowly fixed, perhaps reflecting the range of emotions and contexts in which humans employ them. The evolutionary significance of scream acoustics is discussed.

3aABb3. Discrimination of ripple spectra in a bottlenose dolphin in quiet and after noise exposure. Dmitry Nechaev, Vladimir Popov (Inst. of Ecology and Evolution, Moscow, Russian Federation), Alexander Supin, Mikhail Tarakanov, and Evgeniya Sysueva (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, evgeniasysueva@gmail.com)

The frequency resolving power (FRP) of hearing in quiet and after noise exposure was measured in a bottlenose dolphin using rippled-spectrum test stimuli and noninvasive recording of rhythmic evoked responses (the rate following response, RFR) to ripple phase reversals. Both the test signal and noise had band-limited spectra with the same central frequency; however, the noise had a non-rippled spectrum. The noise level was from -20 to 10 dB re test signal level. The baseline ripple density resolution depended on signal level and was the highest at levels from 80 to 100 dB re 1 μPa. At signal levels both above (up to 130 dB re 1 μPa) and below (down to 80 dB re 1 μPa) the optimal level, the ripple density resolution decreased. The impact of noise was different for different test signal levels. For low test signal levels (70 to 100 dB re 1 μPa), noise decreased RFR magnitude and resolution, whereas for high test signal levels (110 to 130 dB re 1μPa), low-level noise increased RFR magnitude and resolution. [Work supported by the Russian Science Foundation (Project 17-74-20107) awarded to E.V.S.]

3aABb4. Penguin vocalization classifications. Alexandra M. Parshall and Peter M. Scheifele (Univ. of Cincinnati, #31, 2147 Madison Rd., Cincinnati, OH 45208, parshaam@mail.uc.edu)

The present study researches five different penguin species vocalizations who currently residing at the Newport Aquarium in Newport, KY. These penguins—while different species—coexist in the same habitat at the Newport Aquarium. It was purposed that the different species do not "speak the same language" and therefore do not respond to different species of penguin's calls, despite sharing the same enclosure. Therefore, this study aimed at collecting and categorizing the different species' calls and then proceeding to analyze the vocal data to see if any of the species shared similar vocal spectrum similarities. Once collected, the aim of this data is twofold: primarily to classify the different penguin species calls—especially during specific events (i.e., defending a nest, distress, calling for a mate, etc.) and secondarily to analyze the vocal spectral energy of the different species to see if similarities exist.

3aABb5. Analysis of BAER tests of *Tursiops truncatus Montagu* in comparison of captive males to naval trained males in the open ocean. Elizabeth Hanson and Peter M. Scheifele (Univ. of Cincinnati, 3230 Eden Ave., Cincinnati, OH 45342, hansonea@mail.uc.edu)

Do captive Atlantic Bottlenose Dolphins, *Tursiops truncatus Montagu*, have impaired hearing? This study compared the AEP results from captive dolphins to those who are trained by the Navy in the open ocean. The dolphins removed from the wild and relocated to captive environments could have a reduced hearing range as a result of the transfer to an area with potentially increased baseline noise levels. Auditory Brainstem Response (ABR) tests were conducted with underwater acoustic stimuli projection and reception in both the captive and the Navy dolphins. *Tursiops truncatus Montagu* has a range of hearing from 0.1-120 kHz. The hearing of dolphins that were born in captivity was closest to the naval dolphins in the ocean compared to those who were rescued from the wild. Captive dolphins were exposed to a constant 0.1-kHz noise levels for a brief period of time. This may potentially have had an effect on the latency of the peaks I-V and the correlating troughs in their ABR waveform. The results of this study can be used to assess auditory health of marine mammals and to determine the effect of anthropogenic noise levels whether in captive or natural marine mammal environments.

3aABb6. Analysis of bioacoustics and noise pollution in the penguin exhibit at Newport Aquarium. Julie Colaianni and Peter M. Scheifele (Audiol., Univ. of Cincinnati, 3216 Glendora Ave., Cincinnati, OH 45220-2206, colaiaja@mail.uc.edu)

The present study researches the acoustic habitat of the penguin exhibited at Newport Aquarium in Newport, KY by means of measuring environmental noise within their exhibit. Noise exposure has been well studied in humans by using a weighted decibel scale to provide safety guidelines for the level of noise and the maximum time one can be exposed to each level. Therefore, the same principle is being used to observe if these animals are experiencing noise pollution from their own environmental sounds. The absorption coefficients of the materials that make up the habitat were also taken into consideration during data collection. The aim of this research is to gather calculations for the time it takes for the sound pressure level to reduce by 60dB (RT60) after periods of vocalization by the colony of penguins and to determine if these reverberation values reach the peak sensitivity of hearing thresholds for them.

WEDNESDAY MORNING, 15 MAY 2019 NUNN, 9:00 A.M. TO 11:55 A.M.

Session 3aBA

Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Interaction of Light and Ultrasound I

E. Carr Everbach, Cochair
Engineering, Swarthmore College, 500 College Avenue, Swarthmore, PA 19081

Ronald A. Roy, Cochair
Dept. of Engineering Science, University of Oxford, 30 Edgeway Road OX3 0HD, United Kingdom

Invited Papers

9:00

3aBA1. Photoacoustic imaging in cancer medicine and research: Systems, results and future directions. Jeffrey C. Bamber (Joint Dept. of Phys., Inst. of Cancer Res. and Royal Marsden Hospital, 6 Sandbourne Ave., Merton Park, London SW19 3EN, United Kingdom, jeffrey.bamber@physics.org)

This paper reviews selected previous work in photoacoustic imaging conducted at the Institute of Cancer Research or with collaborators and considers directions for future research. For flexible introduction as an additional mode to clinical freehand ultrasound scanning, a multispectral epiphotoacoustic system was built by adapting a Zonare z.oneUltra^{TM}. Early clinical findings from photoacoustic imaging of breast tumours were consistent with contrast MRI although also provided evidence that photoacoustic image interpretation and penetration were limited by clutter generated by photoacoustic emissions from regions at or near the skin surface. Schemes for improving photoacoustic signal-to-clutter ratio were therefore explored. These have included deformation compensated averaging, spatial coherence processing, photoacoustic contrast agents, and localised vibration tagging (LOVIT). LOVIT (implemented using acoustic radiation force) was particularly successful, tripling penetration depth in breast-mimicking phantoms. It has also been implemented on a Verasonics^{TM} platform and extended using a 3 + 3 interleaved comb-push method, which further reduces the clutter and improves the frame rate. Multidimensional and other extensions to LOVIT remain possible. Meanwhile, work at other institutions has explored various propagation model-based schemes. Most of the clutter reduction methods are complementary in nature, and a combined approach therefore represents a worthwhile direction for the future work.

9:20

3aBA2. Synergy of light and sound for deep-tissue optical imaging and focusing. Puxiang Lai (Biomedical Eng., Hong Kong Polytechnic Univ., ST412, BME, Hung Hom, Kowloon, Hong Kong, puxiang.lai@polyu.edu.hk)

Light and sound are the two most dominant mechanisms with which we naturally perceive this world. Both have their significant impacts yet with inherent limitations. For example, sound is insensitive to soft tissue functional changes, and light is strongly scattered in tissue, resulting in a trade-off between penetration depth and resolution. Here, we present our most recent research efforts of exploiting the synergy of light and sound to overcome this limitation. Specifically, photoacoustics converts diffusive photon into non-scattering ultrasonic waves, enabling a high-contrast sensing of optical absorption with ultrasonic resolution in deep tissue, overcoming the optical diffusion limit from the signal detection perspective. The generation of photoacoustic signals, however, is still throttled by the attenuation of photon flux due to the strong diffusion effect of light in tissue. Therefore, wavefront shaping is introduced, so that multiply scattered light could be manipulated so as to retain optical focusing or sufficient photon flux even at depths in tissue. We will present the recent development of photoacoustic imaging and optical wavefront shaping in our lab. Potential applications, existing challenges, and further improvement are also discussed.

9:40

3aBA3. Irreversible shifts in optical autofluorescence spectra applied to the assessment of thermal lesion formation under high intensity focused ultrasound. Shamit Shrivastava (Dept. of Eng. Sci., Oxford Univ., Oxford, Oxfordshire OX1 3PJ, United Kingdom, shrivastava.shamit@gmail.com), E. Carr Everbach (Eng., Swarthmore College, Swarthmore, PA), Jason L. Raymond, and Ronald Roy (Eng. Sci., Oxford Univ., Oxford, United Kingdom)

High-intensity focused ultrasound (HIFU) is often used to create lesions, or regions of tissue destruction due to heating and cavitation activity, most often in tumors or other diseased tissues. However, the acoustic properties of tissues denatured by heat are not very different from those of untreated tissue, making lesion detection and quantification difficult by ultrasound alone. Autofluorescence refers to the broadband emission of light within materials that are stimulated by narrowband incident light, typically from a laser. It is a common characteristic of lipids, proteins, and other biomolecules, and the "autofluorescence spectrum" is a function of the state of the material. We have examined irreversible shifts in the dominant autofluorescence spectra of proteins denatured by HIFU heating. These shifts appear to be related to protein conformational changes due to denaturation. We report on the feasibility of using optical autofluorescence as a means of quantifying *in vitro* lesion formation by HIFU for optically accessible tissues.

10:00–10:15 Break

10:15

3aBA4. Spectroscopic photoacoustic imaging for cardiovascular interventions. Sophinese Iskander-Rizk (Biomedical Eng., Erasmus MC, WYTEMAWEG 80, Rotterdam 3015CN, The Netherlands, s.iskander-rizk@erasmusmc.nl), Pieter Kruizinga (Neurosci., Erasmus MC, Rotterdam, The Netherlands), Min Wu (Biomedical Eng., TU Eindhoven, Rotterdam, The Netherlands), Antonius F. W. van der Steen, and Gijs Van Soest (Biomedical Eng., Erasmus MC, Rotterdam, The Netherlands)

Alongside ultrasound imaging, photoacoustic imaging (PA) augments the imaging system with tissue composition information. Indeed, PA primarily maps tissue optical absorption and thus can reveal tissue composition. Localizing and identifying specific biomolecules can be very useful for both diagnostic imaging and evaluation of the treatment effect in various applications. PA imaging can for instance discern lipid-rich atherosclerotic plaques from fibrotic ones. It can also distinguish highly vascularized or highly oxygenated tissue from hypoxic areas in tumors. However, multiple parameters affect the PA signal received, rendering a direct mapping from signal received to absorption challenging. For instance, at selected wavelengths, multiple chromophores may contribute to the signal observed. In order to separate the different chromophores, judicious spectral tuning of imaging wavelengths can help. We call this technique spectroscopic photoacoustic imaging (sPA). Based on sPA imaging, we were capable to super-localize sources down to 1/20th of the imaging system point spread function (PSF). sPA can also improve the imaging specificity and sensitivity of targeted tissue features. In this talk, we showcase the many benefits of sPA imaging for image enhancement. We also discuss challenges and feasibility to integrate in surgical tools for minimally invasive interventions. In particular, we focus on how sPA studies can take the specific problem of visualization of RF ablation for atrial fibrillation from bench to bedside.

10:35

3aBA5. Towards photoacoustic sensing and mapping of neuroelectrical and hemodynamic activity. Parag V. Chitnis (Dept. of Bio-Eng., George Mason Univ., 4400 University Dr., 1G5, Fairfax, VA 22032, pchitnis@gmu.edu)

Understanding the neurological function and disorders is an enduring challenge, particularly because the brain activity involves a diverse set of chemical, ionic, and electrical interactions spanning a wide range of spatial scales from microns to several centimeters. Recording these time-varying dynamics in intact, mammalian brain provides an insight into how signaling is processed in neural networks and how these signals modulate physiological function. Optical-fluorescence techniques have emerged as tools of choice for the imaging neuronal activity. Despite significant advances in fluorescent voltage and calcium reporters, these methods are limited to penetration depths of less than 1 mm. One viable alternative to overcome the depth limitation is photoacoustic sensing, which relies on absorption of light and subsequent thermoelastic generation of ultrasound. Photoacoustic methods provide spectroscopic specificity to endogenous and exogenous chromophores, but the molecular information is relayed to the sensor acoustically, which is not as susceptible to scattering in tissue as light. We will present on ongoing efforts to develop (1) PA-based voltage reporters, (2) photoacoustic imaging of the voltage and hemodynamic activity at micro- and meso-scales, and (3) a deep learning framework for achieving super-resolution (exceeding the diffraction limit of ultrasound) photoacoustic tomography of the brain. [Partially supported by NIH-1R21EY023012.]

10:55

3aBA6. Super-resolution approaches in photoacoustic imaging. Bastien Arnal, Sergey Vilov, Guillaume Godefroy (LiPhy, Université Grenoble Alpes, CNRS, Grenoble F-38000, France, bastien.arnal@univ-grenoble-alpes.fr), and Emmanuel Bossy (LiPhy, Université Grenoble Alpes, CNRS, Saint-Martin d'Hères, France)

The resolution of photoacoustic imaging (PAI) is limited at depths by the diffraction limit. Several ways have been introduced to achieve super-resolution. In the context of imaging the vasculature, the presence of flow can be exploited in two regimes, distinct by the concentration of flowing absorbing particles. In the high concentration regime, we proposed to exploit the absorption fluctuation caused by flowing absorbers by analyzing nth-order statistics of temporal signal fluctuations. In the low concentration regime, when absorbers appear one-by-one in each acoustic resolution spots, the localization microscopy technique can be adapted to our problem. While these two methods improve the resolution greatly, their cost is to reduce temporal resolution, because of the need to record thousands of images. Supposing the knowledge of the PSF (point spread function) of the imaging system, it is possible to the regain temporal resolution. After the simulation of the forward model, the imaged object can be recovered by solving a minimization problem. We will show that adding a sparsity constraint to this problem can enhance the resolution. These techniques have been investigated in both simulations and experiments in microfluidic channels. Such super-resolution approaches bring the optical contrast at depth closer to the cellular level.

11:15

3aBA7. Sono-photoacoustic theranostics using phase-changing contrast agents. David S. Li, Kacper Lachowski (Dept. of Chemical Eng., Univ. of Washington, 105 Benson Hall, Box 351750, Seattle, WA 98195, dsli@uw.edu), Ivan Pelivanov (BioEng., Univ. of Washington, Seattle, WA), Thomas Matula (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), Matthew O'Donnell (BioEng., Univ. of Washington, Seattle, WA), and Lilo Pozzo (Chemical Eng., Univ. of Washington, Seattle, WA)

Phase-change contrast agents are low boiling point liquid perfluorocarbon droplets that can be vaporized to form larger microbubbles. Droplet vaporization can be used for both imaging and therapy. Sono-photoacoutics is a non-linear imaging method that uses simultaneous optical and acoustic pulses to activate phase-change contrast agents. By combining photothermal heating from a laser pulse and the negative pressure from an acoustic pulse, lower droplet activation thresholds are achieved than from either source alone. In this study, we demonstrate in an *in vitro* model that sono-photoacoutics activation of polypyrrole coated perfluorocarbon droplets can be used to disrupt 2-cm long fibrin clots to restore flow. Polypyrrole coated droplets under 200 nm in diameter were introduced upstream and allowed to diffuse into the clot. A 1.24-MHz single element transducer coaxially aligned with a 1064-nm pulsed laser was used to scan the clot, activating any agents within the clot. Agent activation was quantified using passive cavitation detection, while flow was monitored using a digital balance. Our results show that the droplets can freely diffuse into fibrin clots and SPA activation can restore approximately 25% of the flow.

11:35

3aBA8. Light, sound, nanobubbles: New approach to contrast-enhanced ultrasound and photoacoustic imaging. Stanislav Emelianov (ECE and BME, Georgia Inst. of Technol., 777 Atlantic Dr., Atlanta, GA 30332-0250, stas@gatech.edu)

To overcome the most significant deficiencies of conventional and contrast-enhanced ultrasound imaging—low contrast and large size of microbubbles, we introduced new class of contrast agents—nanometer scale particles that are capable of escaping vascular compartments, penetrating into tissue, and then, once they reach the target site, generating sufficient ultrasound and photoacoustic contrast upon user-controlled optical activation. These multimodal contrast agents—phase-change perfluorocarbon nanodroplets and plasmonic nanoparticles covered by azide compounds—are stable at physiological temperatures, biocompatible, and monodisperse in size. Given the unique properties of the particles, our approach to image these particles is drastically different and is based on ultrasound read-out of the optically induced temporal changes. Specifically, time-varying ultrasound signals exhibited by the nanoparticles versus the static background are used to reconstruct a high contrast, background-free image of the contrast agent. Furthermore, these particles allow for multiplexed molecular imaging by permitting user-controlled triggering of distinct color-coded populations of contrast agents via tuning of the incident laser irradiation to match peak optical absorption of the particles. Finally, nanoparticles may also contain therapeutic cargo and thus can be used for controlled drug delivery and release. This presentation, via examples, will discuss diagnostic imaging and image-guided therapy using the gas-generating nanoparticles.

Session 3aCA

Computational Acoustics, Biomedical Acoustics, Physical Acoustics, Underwater Acoustics, Signal Processing in Acoustics and Structural Acoustics and Vibration: Finite Difference Time Domain Method Across Acoustics

Michelle E. Swearingen, Cochair
U.S. Army ERDC, Construction Engineering Research Laborartory, P.O. Box 9005, Champaign, IL 61826

Jennifer Cooper, Cochair
Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Rd, Mailstop 8-220, Laurel, MD 20723

Subha Maruvada, Cochair
U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993

Invited Paper

9:00

3aCA1. Applications of finite-difference time-domain for architectural acoustics consulting. Laura C. Brill and John T. Strong (Threshold Acoust., 141 W. Jackson Blvd., Ste. 2080, Chicago, IL 60604, jstrong@thresholdacoustics.com)

Acoustics consultants have many tools in their arsenal to evaluate and design rooms and architectural elements; the computational resources available to this point have made the use of wave-propagation models impractical for the common user. Threshold acoustics has found it both useful and now computationally feasible to supplement more traditional, geometric analysis with the simulation of wave-propagation using finite-difference time-domain (FDTD). Our group has developed first-order leapfrog FDTD routines in MATLAB for simulating wave propagation in an isotropic medium in two- and three-dimenision with perfectly matched layers being the boundary condition. The placement of solid elements within the test space allows analysis of arbitrary geometries. For additional computational power, our group has utilized GPU computing clusters available through Amazon Web Services accessed directly through MATLAB. Our method is based on the simulation of an impulse response and subsequent analysis of the impulse response consistent with traditional *in situ* testing methods. Applications to date include analysis of the scattering behavior of acoustically shaped surfaces and evaluation of the array behavior of architectural reflector panels.

Contributed Papers

9:20

3aCA2. Small room auralizations: Investigating hybrid methods of acoustic simulations utilizing wave field synthesis. E. K. Ellington Scott and Jonas Braasch (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, scotte3@rpi.edu)

Auralizations have become a beneficial tool in acoustic design of performance spaces, but nearly all research has been limited to Western classical music venues. Other venues, for example, jazz clubs, pose very different auralization challenges because their dimensions are typically smaller, and pure geometrical methods can no longer be applied. With a dramatic increase in computational performance, an effort has been placed in integrating geometric and wave-based models in auralization simulations that are suitable to auralize smaller venues. Finite difference time domain methods are applied for low-frequency numerical analysis of the Dizzy's Club Coca-Cola and The Village Vanguard. This research endeavors to establish a method in identifying the crossover frequency between the geometric and wave-based models, as it pertains to small room acoustics to recreate the sound of jazz venues using the wave field synthesis. The 128-channel wave field synthesis system of Rensselaer's Collaborative Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab) is used to perceptually evaluate the hybridized auralizations. Room acoustical measurements are obtained within the rendered sound field and compared to the measurements of the original locations for an objective analysis.

9:35

3aCA3. Finite difference time domain ray-based modelling of acoustic scattering for target identification and tracking. Grant Eastland (Test & Evaluation, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

The Finite Difference Time Domain (FDTD) method has provided a powerful technique for modelling and simulation of solutions of a variety of acoustics problems. The purpose of this investigation is to present work on the development of time-domain models of acoustic scattering from targets near a flat pressure-release boundary for use in identification and tracking from a moving receiving platform. The timing from the acoustic source to reception is dependent on the location dependent sound speed profile and the specular scattering points on the target. There can be multiple specular points revealed on the target because of interactions with the flat boundary, each can be modelled with the FDTD method and providing the correct path corrections using the same method to account for the variation in propagation path brought on by the sound speed profile of the propagating environment.

9:50

3aCA4. Acoustic wave scattering from dynamic rough sea-surfaces using the finite-difference time-domain method. Alex Higgins and Martin Siderius (Elec. & Comput. Eng., Portland State Univ., 1900 SW 14th Ave., Ste. 25-01, Portland, OR 97201, higginja@ece.pdx.edu)

Numeric models for underwater acoustic propagation typically assume the sea-surface to be either perfectly smooth or rough but "frozen" in time. For long sonar signals on the order of tens of seconds, the sea-surface can interact at many different wave displacements over its duration. This causes anomalies in the received signal which introduces additional transmission losses and Doppler effects. The impact of including roughness and motion of the sea-surface on sonar systems is investigated using the finite-difference time-domain (FDTD) method. The FDTD method is a numeric technique that is well suited for modeling boundary roughness and motion. This is due to its ability to directly configure complex boundary conditions in the surrounding simulation grid and full pressure wave propagation in the time-domain. The rough, time-evolving sea-surface is modeled using a Pierson-Moskowitz (PM) frequency spectrum which is simple to implement and defined using just wind speed and direction. The results from FDTD simulations of static rough sea-surfaces are compared to a previously established integral solution method to evaluate the validity of the approach. Agreement is also demonstrated for FDTD simulations of a dynamic rough sea-surface and a theoretic statistical model. [Work supported by the Office of Naval Research.]

10:05–10:20 Break

10:20

3aCA5. Time-domain simulations of sound propagation near a ground surface in comparison with the ANSI impedance measurement models. Z. C. Zheng, Junjian Zhang (Aerosp. Eng., Univ. of Kansas, 2118C Learned Hall, 1530 W 15th St., Lawrence, KS 66045, zzheng@ku.edu), and Guoyi Ke (Mathematics and Physical Sci., Louisiana State Univ. of Alexandria, Alexandria, LA)

Time-domain simulations with the Zwikker-Keston porous-material model are carried out on the geometries specified in the ANSI models for determining the acoustic impedance of ground surfaces (ANSI/ASA S1.18). The comparisons between the simulation results and the results provided in ANSI show very good agreement when the flow resistivity of the ground material is high. The effect of ground roughness is investigated by adding periodic spacing triangular strips on the ground. The numerical method for simulating rough ground surfaces, which combines the time-domain simulation with an immersed-boundary method, is validated by comparing with experimental data in the literature. It is found that when the ground roughness is introduced to the ANSI geometries, the predicted sound level difference between the two microphones in the ANSI geometries tends to shift towards lower frequency ranges with the rough ground, even though the roughness is within the allowed roughness height specified in the ANSI models.

10:35

3aCA6. Application of Elastodynamic Finite Integration Technique (EFIT) to three-dimensional wave propagation and scattering in arbitrary geometries. Sean Raley and Eric A. Dieckman (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, srale1@unh.newhaven.edu)

Over several decades, railroad ultrasonic examination (UE) industry techniques have primarily been developed through simple analytical modeling and experimental approaches. However, with present-day computational capabilities, we can use numerical techniques such as the Elastodynamic Finite Integration Technique (EFIT) to fine-tune systems for complex applications before the fabrication process begins. EFIT is well-established as a useful method in numerical analysis of ultrasonic wave propagation with distinct advantages over the Finite Difference Time Domain method. Several software packages exist that use EFIT as the primary method for simulating the behavior of ultrasonic waves over time in two or three dimensions, but none of them are well-suited for railroad UE R&D. This paper explores the development of a tool developed for this purpose which was designed to: (1) allow for the input of various profile geometries, boundary conditions, and material inclusion geometries (such as a bolt hole in a railroad track); (2) allow for the input of specific ultrasonic impulses from varying emitter designs; and (3) produce verifiable results, as confirmed by experimental measurements.

Invited Paper

10:50

3aCA7. Estimation of parameters quantifying porosity in random porous structures using ultrasonic attenuation: Solving the inverse problem. Omid Yousefian (North Carolina State Univ., Raleigh, NC), Rebekah White (North Carolina State Univ., Chapel Hill, NC), H. T. Banks, and Marie M. Muller (North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

The goal of this study is to estimate the porosity (pore size and density) of numerically simulated random porous three-dimensional structures mimicking simplified geometries of cortical bone using attenuation of ultrasonic waves in the MHz range. To do so, we use a physics-based model derived from the Independent Scattering Approximation (ISA) to mathematically model the attenuation of elastic waves in porous structures as a function of the parameters that we wish to identify: pore size and density. Frequency-dependent attenuation data are generated using three-dimensional finite-difference time-domain simulations in the range of 1–8 MHz in mono-disperse structures with the pore diameter and density ranging from 100 to 200 μm and 20–50 pore/mm^3, respectively. We then solve an ordinary least squares (OLS) inverse problem to recover the pore size and density by minimizing the sum of squared errors between the simulated data and the model prediction. In doing so, we demonstrate that we can estimate with confidence the parameters quantifying porosity using the ISA model given three-dimensional numerically simulated attenuation data.

Session 3aED

Education in Acoustics: Hands-On Demonstrations

L. Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11747

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

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomena. In this session, "Hands-On" demonstrations will be set-up for a group of middle- and high-school students from the Louisville 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 e-mail Keeta Jones (kjones@acousticalsociety.org).

Session 3aMU

Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Polyphonic Pitch Perception and Analysis I

Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

M. Torben Pastore, Cochair
Architectural Acoustics, Rensselaer Polytechnic Institute, 4 Irving Place, Troy, NY 12180

Chair's Introduction—8:00

Invited Papers

8:05

3aMU1. Pitch perception of concurrent high-frequency complex tones. Daniel Guest and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E River Rd., Minneapolis, MN 55455, guest121@umn.edu)

Accurate pitch perception is possible for harmonic complex tones with fundamental frequencies (F0s) in the musical range (e.g., 1.4 kHz) but with all harmonics beyond the putative limits of phase locking. However, it is unknown whether pitch perception in more complex scenarios, such as with concurrent complex tones, is possible using these stimuli. To address this question, we measured (1) F0 difference limens (F0DLs) and (2) target-to-masker ratios (TMRs) required to detect a fixed F0 difference in a mixture of complex tones with low F0s (∼280 Hz) or high F0s (∼1400 Hz). The target tones were filtered to ensure that in the high-F0 case, only harmonics beyond the limits of phase locking were present. Pitch perception was poorer for isolated high-F0 tones than for isolated low-F0 tones and adding a masker complex tone with a geometrically centered F0 impaired performance for both high-F0 and low-F0 tones. The TMRs required to achieve good performance in the presence of two complex tone maskers were higher for high-F0 tones than for low-F0 tones. The results should help determine whether different mechanisms underlie the perception of combinations of complex tones at low and high frequencies. [Work supported by Grants NIH R01 DC005216 and NSF NRT-UtB1734815.]

8:25

3aMU2. Multi-pitch spike representation using a Finite-Difference Time Domain (FDTD) cochlear model. Rolf Bader (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

The spike representation of multi-pitch sounds leaving the cochlear is discussed using a Finite-Difference Time Domain (FDTD) physical model of the cochlear with musical sounds as input. Previously, the model has lead to a new pitch theory, where it was found that the interspike intervals (ISI) of the fundamental periodicity of a single-pitch sound are present at multiple Bark bands in the cochlear. This is due to drop-outs of spikes in Bark bands of higher harmonics. Pitch is therefore represented as the most prominent periodicity within the auditory nerve. In the multi-pitch case, the model shows several cases of such multi-Bark periodicities. With multi-pitch sounds having a residual pitch, the strongest of these multi-Bark cases is that of the residual periodicity. With several pitches perceptually clearly distinguishable one from another, each pitch periodicity appears as a multi-Bark periodicity. Still with sounds where the single pitches are not so clear and separating them needs musical training and intense and multiple listening of the sound, the multi-Bark cases are blurred. Therefore, the model represents the perception of multi-pitch sounds during immediate perception and identification of multiple pitches needing musical training is no longer present at this low-level sound representation at the cochlear output, as expected.

8:45

3aMU3. Spectro-temporal templates unify the pitch of resolved and unresolved harmonics. Shihab Shamma and Kelsey J. Dutta (Univ. of Maryland, 2202 A.V. Williams, College Park, MD, sas@isr.umd.edu)

Pitch is a fundamental attribute in auditory perception that is involved in source identification and segregation, music, and speech understanding. When harmonics are well-resolved, the induced pitch is usually salient and precise; however, when the harmonics are not completely resolved, the pitch percept becomes less salient and poorly discriminated. Previous models relying on harmonic spectral templates have been able to account fully for the pitch of the resolved but not of the unresolved harmonics. I will describe a biologically motivated model of templates that combine both spectral and temporal cues to estimate both pitch percepts. Specifically, the pitch of unresolved harmonics is estimated through bandpass filters implemented by resonances in the dendritic trees of neurons in the early auditory pathway. It is demonstrated that organizing and exploiting such dendritic tuning can arise spontaneously even in response to white noise. We show how these temporal cues become integrated with those of spectrally resolved harmonics, effectively creating spectro-temporal harmonic templates for all pitch percepts. We finally discuss how this approach can account for all major monaural pitch percepts, as well as pitch percepts evoked by dichotic binaural stimuli.

9:05

3aMU4. Autocorrelation models of pitch perception: In memory of ray meddis. William Yost (Spatial Hearing Lab, ASU, P.O. Box 870102, Tempe, AZ 85287, william.yost@asu.edu) and Roy D. Patterson (Physiol., Univ. of Cambridge, Cambridge, United Kingdom)

About 60 years ago, Licklider (1951, *Experientia*) described a qualitative model of autocorrelation that appeared to explain complex pitch perception. However, it was almost 40 years later before, Slaney and Lyon (1990, *ICAASP*) and Meddis and Hewitt (1991, *JASA*) incorporated autocorrelation into computational auditory models to test Licklider's model of pitch perception. The Meddis and Hewitt frontend combined a level-dependent gammatone filterbank with Meddis' (1986, *JASA*) inner hair-cell model. The "autocorrelograms" it produced revealed the temporal regularity of simulated auditory-nerve interspike intervals that occur in response to complex stimuli. The model was/is remarkably successful in accounting for a large set of complex pitch perception data. The model emphasizes the temporal structure of the neural information produced by a complex sound, in contrast to models that use a simple spectral representation of the resolved harmonics of a complex sound. Today, those who model complex pitch perception either use an autocorrelation-like approach or argue why such an approach does not work. The Meddis-Hewitt model is almost always considered in these modeling efforts. This presentation will describe the Meddis-Hewitt model and discuss its lasting effect on the study of complex pitch perception. [Work supported by NIDCD and Facebook Reality Labs grants to WAY.]

9:25

3aMU5. Fundamental frequency (F0) discrimination of one complex tone in the presence of another: The role of excitation pattern and temporal fine structure cues. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

The perceptual segregation of simultaneous harmonic complex tones depends partly on differences in the fundamental frequency (F0) between those tones. Human listeners have some ability to group together harmonics of a given F0 and segregate them from harmonics of a different F0. This paper reviews studies of the ability to detect changes in F0 of one complex tone in the presence of another complex tone with the same or a different mean F0. The studies include conditions where the harmonics in the complex tones would have been resolved, partially resolved, or completely unresolved. Modeling using excitation patterns and "summary autocorrelation" and "stabilized auditory image" models suggests that while excitation-pattern cues are useful for complex tones with resolved harmonics, the use of temporal fine structure (phase locking) information is required to account for the small F0DLs obtained when harmonics are barely, if at all, resolved.

9:45

3aMU6. Spectral and temporal models of human pitch perception with mixtures of three concurrent harmonic complexes. Jackson Graves (Département d'Études Cognitives, École Normale Supérieure, 29, rue d'Ulm, Paris 75005, France, grave276@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

In music and other everyday situations, humans are often presented with three or more simultaneous pitches, each carried by a harmonic complex tone, but few empirical or theoretical studies have addressed how pitch is perceived in this context. In three behavioral experiments, mixtures of three concurrent complexes were filtered into a single bandpass spectral region, and the relationship between the fundamental frequencies and spectral region was varied in order to manipulate the extent to which harmonics were resolved either before or after mixing. Listeners were asked to discriminate major from minor chords (Experiment 1) or to compare the pitch of a probe tone to that of a target embedded in the mixture (Experiments 2 and 3). In all three experiments, listeners performed above chance even under conditions where traditional rate-place models would not predict individually resolved components. Human behavioral results were compared to predictions from two classes of pitch model: a rate-place model using harmonic template matching and a temporal model using summary autocorrelation. Predictions from a combined model, using both rate-place and temporal information, were more accurate than predictions from either model alone, suggesting that humans may integrate these two kinds of information. [Work supported by NIH grant R01DC005216.]

10:05

3aMU7. Perceptual fusion of musical notes suggests universal representations of dissonance despite culture-dependent aesthetic associations. Malinda J. McPherson (Div. of Medical Sci., Harvard Univ., MIT Bldg. 46-4078, 43 Vassar St., 46-4078, Cambridge, MA 02139, malindamcpherson@g.harvard.edu), Sophia E. Dolan (Wellesley College, Wellesley, MA), Tomas Ossandon, Joaquin Valdes (Dept. of Psychiatry, Pontificia Universidad Católica de Chile, Santiago, Chile), Eduardo A. Undurraga (Escuela de Gobierno, Pontificia Universidad Católica de Chile, Santiago, Chile), Nori Jacoby (The Ctr. for Sci. and Society, Columbia Univ., New York, NY), Ricardo Godoy (Heller School for Social Policy and Management, Brandeis Univ., Waltham, MA), and Josh McDermott (Dept. of Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA)

Music varies enormously across cultures, but some traits are widespread. Cross-cultural consistency in music could be driven by universal perceptual mechanisms adapted to natural sounds, but supporting evidence has been circumstantial due to the dearth of cross-cultural research. Here, we explore whether such perceptual mechanisms impose universal similarity relations on musical structure, potentially dissociating from culture-specific aesthetic judgments about music. We measured one possible signature of these similarity relations—the extent to which concurrent notes are perceived as a single sound—in members of a small-scale Amazonian society and Western listeners. We also measured aesthetic responses to the same stimuli. Unlike Westerners, Amazonian listeners were aesthetically indifferent to whether note combinations were canonically consonant (with aggregate frequency spectra resembling the harmonic series). However, Amazonians were nonetheless more likely to hear consonant combinations as a single sound, with fusion judgments that qualitatively resembled those of Western listeners. Thus, even in a culture with little exposure to Western harmony, reliance on harmonic frequency relations for sound segregation evidently induces consistent perceptual structure in note combinations. The results suggest that perceptual mechanisms for representing music can be shared across cultures, even though the perceptual equivalences that result give rise to culture-specific aesthetic associations.

10:25–10:40 Break

Contributed Paper

10:40

3aMU8. Polyphonic pitch perception in rooms using deep learning networks with data rendered in auditory virtual environments. Nathan Keil, David A. Dahlbom (Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, keiln@rpi.edu), Jeremy Stewart, Matthew Goodheart, Curtis Bahn, Mary Simoni (Arts, Rensselaer Polytechnic Inst., Troy, NY), Michael Perrone (Thomas J. Watson Res. Ctr., IBM, Yorktown Heights, NY), and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

This paper proposes methods for generation and implementation of uniform, large-scale data from auralized MIDI music files for use with deep learning networks for polyphonic pitch perception and impulse response recognition. This includes synthesis and sound source separation of large batches of multitrack MIDI files in non-real time, convolution with artificial binaural room impulse responses, and techniques for neural network training. Using ChucK, individual tracks for each MIDI file, containing the ground truth for pitch and other parameters, are processed concurrently with variable Synthesis ToolKit (STK) instruments, and the audio output is written to separate wave files in order to create multiple incoherent sound sources. Then, each track is convolved with a measured or synthetic impulse response that corresponds to the virtual position of the instrument in the room before all tracks are digitally summed. The database now contains the symbolic description in the form of MIDI commands and the auralized music performances. A polyphonic pitch model based on an array of autocorrelation functions for individual frequency bands is used to train a neural network and analyze the data [Work supported by IBM AIRC grant and NSF BCS-1539276.]

10:55

3aMU9. Matching pitch gestures to visual trajectories as a function of acoustic scales. Sven-Amin Lembke (Music, Technol. and Innovation - Inst. for Sonic Creativity (MTI2), De Montfort Univ., Clephan Bldg., Rm. 00.07b, Leicester LE1 9BH, United Kingdom, sven-amin.lembke@dmu.ac.uk)

Auditory gestures are commonly used in music and sonification, although little is known about their perceived shape. A listening experiment aimed to study how pitch-based gestures that rely on different acoustic scales are perceived and matched to analogous visual trajectories. Four scales were studied, varying in curvature on a linear time-vs-frequency plane: linear frequency in Hz, equivalent-rectangular-bandwidth (ERB) rate, an exponential function twice the curvature of ERB rate, and a logarithmic function with inverse curvature to ERB rate. The pitch-glide stimuli were based on filtered noise and spanned two octaves. These glides were tested below and above a 1-kHz anchor, ascending and descending in pitch, and for two durations, all serving as independent variables (IVs). An interactive visual trajectory served as the dependent variable. Participants matched the visual shape to the perceived auditory gesture, along a continuum from exponential to logarithmic curvatures, including a straight line. Only three of the four acoustic scales were compared and rated in a single trial, which allowed to assess contextual bias across all possible combinations. Preliminary data suggest that pitch glides based on the ERB rate match a straight line best, while linear-frequency glides are perceived as moderately logarithmic trajectories. These two scales, however, exhibit some contextual variability, whereas these trends seem robust across the investigated IVs, intra-subject consistency appears to strongly depend on listening expertise.

11:15

3aMU10. Correlation-based temporal model of pitch multiplicity. Peter Cariani (Hearing Res. Ctr., Boston Univ., 629 Watertown St., Newton, MA 02460, cariani@bu.edu)

Global temporal models for pitch analyze population-wide, all-order interspike interval distributions of the entire auditory nerve to accurately predict almost all known F0-pitch phenomena. Population-interval distributions (PIDs) are neural representations that resemble log-frequency scaled, half-wave rectified summary autocorrelation functions (SACF) of stimuli. PIDs produced by periodic stimuli show regular patterns of lag peaks associated with subharmonics of harmonics, with major peaks at n/F0, whereas early global temporal models chose the highest PID peak to estimate one, dominant pitch, later models compared average interval-densities of different sets of F0-related PID peaks to estimate relative pitch saliences, all saliences above a threshold being audible. However, this method inherently generates octave confusions amongst related subharmonics. Using Pearson correlation coefficients between PIDs and periodicity-related lag patterns (lags at n/F0) to estimate pitch saliences obviates the octave ambiguity problem and lag-weightings. The results from recent simulations using the Zilany-Bruce-Carney ANF model to estimate the pitches heard for dyads and triads of complex harmonic tones will be presented. For triadic chords (major, minor, suspended, augmented, and diminished), the model estimates relative saliences of F0s of notes, fundamental basses, and individual harmonics). Correlation salience values indicating relative pitch strengths (pitch stabilities) generally comport with music theoretic rankings for these chords.

11:35

3aMU11. Human-like pitch perception mechanisms in marmoset monkeys. Xindong Song, Yueqi Guo, Michael Osmanski, and Xiaoqin Wang (Biomedical Eng., Johns Hopkins Univ., 720 Rutland Ave., 412 Traylor Bldg., Baltimore, MD 21205, songxindong@jhmi.edu)

The perception of the pitch of harmonic complex sounds is a crucial function of human audition, especially in music and speech processing. Whether the underlying mechanisms of pitch perception are unique to humans, however, is unknown. Based on estimates of frequency resolution at the level of the auditory periphery, psychoacoustic studies in humans have revealed several primary features of central pitch mechanisms. It has been shown that (1) the pitch strength of a harmonic tone is dominated by resolved harmonics; (2) pitch of resolved harmonics is sensitive to the quality of spectral harmonicity; and (3) pitch of unresolved harmonics is sensitive to the salience of temporal envelope cues. Here, we show that, for a standard musical tuning fundamental frequency of 440 Hz (ISO 16), the common marmoset (*Callithrix jacchus*), a New World monkey with a hearing range similar to that of humans, exhibits all the primary features of central pitch mechanisms demonstrated in humans. Thus, marmosets and humans may share similar pitch perception mechanisms, combined with previous findings of a specialized pitch processing region in both marmoset and human auditory cortex, suggesting that these mechanisms may have emerged early in primate evolution.

Session 3aPA

Physical Acoustics, Engineering Acoustics, and Biomedical Acoustics: Acoustofluidics I

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Invited Papers

8:30

3aPA1. Acoustofluidics: Merging acoustics and microfluidics for biomedical applications. Tony Jun Huang (Pratt School of Eng., Duke Univ., 144 Hudson Hall, Box 90300, Durham, NC 27708, tony.huang@duke.edu)

The past two decades have witnessed an explosion in lab-on-a-chip research with applications in biology, chemistry, and medicine. Recently, a new lab-on-a-chip frontier has emerged, joining acoustics with microfluidics, termed acoustofluidics. Here, we summarize our recent progress in this exciting field and show the depth and breadth of acoustofluidic tools for biomedical applications through many unique examples, including exosome separation, cell-cell communication studies, three-dimensional bioprinting, circulating tumor cell isolation and detection, ultra-high-throughput blood cell separation, high-precision micro-flow cytometry, and portable fluid manipulation systems. These acoustofluidic technologies are capable of delivering high-precision, high-throughput, and high-efficiency cell/particle/fluid manipulation in a simple, inexpensive, cell-phone-sized device. More importantly, the acoustic power intensity and frequency used in these acoustofluidic devices are in a similar range as those used in ultrasonic imaging, which has proven to be extremely safe for health monitoring during various stages of pregnancy. As a result, these methods are extremely biocompatible, i.e., cells and other biospecimens can maintain their natural states without any adverse effects from the acoustic manipulation process. With these advantages, acoustofluidic technologies meet a crucial need for highly accurate disease diagnostics (e.g., early cancer detection and monitoring of prenatal health) and effective therapy (e.g., transfusion and immunotherapy).

9:00

3aPA2. Living probes as calibration standards for acoustic microfluidics. Minji Kim (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, St. Louis, MO), Emma Huff (Biomedical Eng., Washington Univ. in Saint Louis, St. Louis, MO), Philip Bayly, and J. Mark Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Urbauer Hall, Rm. 319, St. Louis, MO 63130, meachamjm@wustl.edu)

Acoustic microfluidics is promoted as an enabling technology for numerous applications in medicine and biology; however, adoption of these solutions in clinical and industrial settings is hampered by inconsistent performance and poor reproducibility. Though computational modeling and laboratory-scale demonstrations anticipate significant advantages for acoustofluidic unit operations including mixing, sorting/separation, and isolation, few technologies have realized this potential. A lack of standard tools and methods for assessing and comparing device performance represents a critical barrier to progress in the field. Here, we introduce a living probe that allows accurate and dynamically responsive measurement of the acoustic pressure within a device. Motile unicellular alga *Chlamydomonas reinhardtii* (CR) probe their environment and naturally swim against an imposed force field to fill complicated shapes. Steady-state distributions of swimming cells can be related to the field shape and strength to more completely describe the pressure field (versus passive particles that reach terminal distributions at nodal locations). Significantly, CR cells continuously respond to their environment, which enables real-time observation of the system response to varying operating conditions (e.g., frequency and/or drive voltage). We present the results that demonstrate correlation of CR cell distributions with pressure fields in simple one- and two-dimensional shapes, as well as more complex architectures.

9:30

3aPA3. Extraordinary manipulations of particles by acoustic radiation forces and torques. Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu)

Particle manipulations using radiation forces and torques produced by acoustic waves have been found valuable applications in biomedical and material engineering. Recent work suggests extraordinary manipulations of particles associated with direction reversals of forces and torques by some specific sound beams. When considering the three-dimensional acoustic radiation force, the momentum view enlightens into the superposition and coupling of manipulations by multiple beams and the force dependence on scattering functions [L. Zhang, *JASA* **144** (1), 443–447 (2018)]. It is efficient to apply the superposition to consider forces and torques on particles moved away from the equilibrium trapping position [L. Zhang, *JASA* **143**(5), 2796–2800 (2018)]. The superposition reveals how the radiation torque associated with energy absorption can spin an object around its center of mass in a direction reversed with respect to the wave vortex's handedness [L. Zhang, *Phys. Rev. Appl.* **10**(3), 034039 (2018)]. These insights into extraordinary manipulations are of interest for advancing the flexibility of particle manipulations by acoustic tweezers and to the field of acoustofluidics.

10:00–10:15 Break

10:15

3aPA4. New developments in acoustofluidics: Understanding and utilization from atomization to nanofluidics. James Friend (Mech. and Aerosp. Eng., Structural and Mech. Eng., Univ. of California, San Diego, 345F, M.S. 411 Gilman Dr., La Jolla, CA 92093, jfriend@eng.ucsd.edu)

We report new developments in understanding and utilizing acoustofluidics, including atomization devices and physics, interesting new phenomena observed in nanochannels, and entirely new directions of research. In our atomization effort, we have found an absence of bulk turbulence previously posited responsible for capillary wave phenomena that give rise to atomization and find a complex combination of caustics and nonlinear interfacial deformation to be responsible instead. These phenomena are observed on our completely portable micro-scale atomizer devices potentially useful in pulmonary applications. In extending our previous efforts in nanoslit acoustofluidics, we examine droplet manipulation in two-dimensional nanochannel arrays and observe transport at up to 0.1 m/s. Finally, we describe methods and the underpinning physics to integrate acoustofluidics into energy storage systems for a step change in their performance.

Contributed Papers

10:45

3aPA5. Calibration of an ultrasonic transducer in a standing wave field. Krishna N. Kumar, Tyler Campbell, Jack Saloio, Kedar C. Chitale (Res. & Development, FloDesign Sonics, Inc., 380 Main St., Wilbraham, MA 01095, k.kumar@fdsonics.com), and Bart Lipkens (Res. & Development, FloDesign Sonics, Inc., Springfield, MA)

Acoustofluidics is one of the emerging technologies for cell sorting. Most of the acoustofluidics platforms use an acoustic standing wave field for the separation. The calibration of the standing wave field is a challenging task. In the past, there have been few studies to quantify the standing wave indirectly using particle tracing in the field. In the present work, we attempt to calibrate the standing wave field using acoustic techniques. Hydrophones are used to find the pressure in an acoustic field. They are designed and constructed in a way that they do not alter the field. Most of the commercial hydrophone in market is made of PVDF (polyvinylidene fluoride): an acoustically transparent material. When a hydrophone is inserted in a standing wave field, the field in front of the tip is no longer a standing wave field as PVDF is acoustically transparent. This was observed in our acoustic chamber also. Here, we propose to use the receiving characteristics of a transducer to quantify the standing wave field. In brief, two identical PZT transducers are used to generate the standing wave field: one as a transmitter and other as a receiver (reflector). The receiving characteristics of the transducer used as a reflector are calibrated using a needle hydrophone. Using the signal recorded by the receiving PZT transducer in the standing wave field and the calibration value, the pressure at the surface of reflecting transducer can be found. For the sake of comparison, the field will be quantified also by the particle tracing method.

11:00

3aPA6. Theory of sound propagation through a fluidic medium with suspended clay and silt particles having the general characteristics of mud. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., P.O. Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Work presented at previous meetings and in various publications for propagation of sound through mud continues with a theory that simultaneously takes into account viscous flow past both suspended clay and silt particles. Why silt particles do not settle under the action of gravity to the bottom of the layer is explained because (1) the clay particles adhere together in a flocculated card-house configuration and (2) the silt particles are trapped in the clay matrix. The conjecture is defended that the clay matrix and the silt particles move in lock-step under the influence of an incident sound wave. Neighboring particles of different sizes move with the same velocity amplitude, although they are subjected to different viscous forces. The theories of Stokes, Happel, and Brenner are used to calculate viscous forces at low frequencies for particles of different shapes, with the clay particles idealized as thin platelets and the silt particles idealized as spheroids with different eccentricities and random orientations. The forces on the clay particles continue to be given by the low frequency approximation for all frequencies of interest, and the deviations from Stokes's low frequency law are taken to be what corresponds to a sphere with the equivalent radius. The size distributions of the particles are taken from existing data. Modified fluid dynamic equations are derived using basic principles. The attenuation is shown to vary as frequency squared at low frequencies and as the square root of frequency for higher frequencies.

Session 3aPP

Psychological and Physiological Acoustics and Speech Communication: Context Effects in Speech Perception I

Christian Stilp, Cochair
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Matthew Winn, Cochair
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Invited Papers

8:05

3aPP1. Speech perception is influenced by the speech rate of both attended and unattended sentence contexts. Hans Rutger Bosker (Max Planck Inst. for PsychoLinguist, P.O. Box 310, Nijmegen 6500 AH, The Netherlands, HansRutger.Bosker@mpi.nl)

Speech perception is influenced by the temporal properties of the surrounding acoustic context. These temporal context effects (TCEs) are contrastive: a heavily reduced target sound [tɛɹ:] is perceived as "long" "terror" when preceded by fast speech but as "short" "tear" in slow speech. I will introduce earlier studies demonstrating that TCEs involve domain-general processes arising early in perception, driven by neural oscillations entraining to the syllabic rhythm of speech. If TCEs arise early in perceptual processing, this raises the question whether TCEs are modulated by selective attention. That is, is speech perception in a "cocktail party" situation influenced by attended speech rates only or also by the speech rate of unattended talkers? In three experiments, participants were presented with two simultaneous context sentences, one in each ear, followed by diotic ambiguous target words. The speech rate of both attended and unattended talkers was found to equally influence target categorization, regardless of whether the attended context was in the same or different voice than the target, and even when participants could watch the attended talker speak. Therefore, TCEs are immune to selective attention, suggesting that TCEs largely operate at a level in the auditory processing hierarchy that precedes attentional stream segregation.

8:25

3aPP2. Distinguishing peripheral and central contributions to speech context effects. Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Acoustic context effects often function in a spectrally contrastive manner and can contribute to achieving perceptual constancy of speech sounds under widely varying conditions, including different room acoustics, different talkers, and different background noises. The neural mechanisms underlying these effects remain unknown, but they are likely to involve within-channel adaptation and across-channel inhibition, similar to that postulated for non-speech auditory context effects, such as auditory enhancement. These mechanisms could be instantiated as early as the cochlea, via efferent circuits, but may also involve higher-level processes. All such mechanisms could in principle be modulated by expectation and attention. Work from our lab has attempted to shed light on the nature and locus of the mechanisms underlying auditory and speech context effects, using a combination of techniques, including behavior, otoacoustic emissions, and electroencephalography (EEG). The results so far suggest that context effects can be somewhat modulated by attention, and may involve stages of auditory processing that extend to subcortical regions, but seem unlikely to involve changes in cochlear mechanics. [Work supported by NIH grant R01DC012262.]

8:45

3aPP3. An individual differences approach to acoustic context effects in speech categorization. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Objects and events in the sensory environment are not perceived in isolation but in the context of surrounding stimuli. This fact has been widely appreciated in the study of speech perception, as demonstrations of these context effects date back more than a half-century. For example, the spectral composition of surrounding sounds can bias categorization of later speech sounds (spectral context effects). Similarly, temporal characteristics of surrounding sounds can bias categorization of target speech sounds (temporal context effects). The literature features many individual reports of listeners experiencing these context effects, but what do these results mean across studies? If a listener exhibited a large context effect in one experiment, is that suggestive of him/her exhibiting a large context effect in a different experiment (i.e., general sensitivity to acoustic context) or not (i.e., results limited to certain stimuli / acoustic domains)? Here, I will take an individual differences approach to spectral and temporal context effects, evaluating their predictive power within and across domains, and how this might inform future work in these areas.

9:05

3aPP4. The changing social context of speech perception: Comparisons across age and time. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Beliefs about talkers' social-group membership affect how listeners perceive their speech: listeners identify phonemes differently depending on whether they believe that the talker was a man or a woman (Strand and Johnson, 1996; Munson, 2011), whether the talker was young or old (Drager, 2011), and whether the talker is gay or straight (Mack and Munson, 2012). These social influences are thought to reflect knowledge of social norms associated with categories such as gender, age, and sexuality. Social norms themselves are subject to change. For example, attitudes toward gay and lesbian people have changed dramatically in the past 15 years. For example, a strong majority of young adults now report having a positive view of sexual minorities. This talk reports on ongoing studies in our lab that have examined whether changes in social norms about gender and sexual orientation are associated with changes in how talkers' gender and sexual orientation affect listeners' speech perception (Obeda and Munson, 2018; Obeda *et al*., 2018). We have found robust differences between archival data from our lab from 2003 and both older and younger listeners tested in 2018. This talk will discuss the implications of these findings for models of speech perception more generally.

Contributed Paper

9:25

3aPP5. Semantic context influences early speech perception: Evidence from electrophysiology. Laura Getz and Joseph C. Toscano (Psychol. and Brain Sci., Villanova Univ., 800 E. Lancaster Ave., Villanova, PA 19085, laura.getz@villanova.edu)

An unresolved issue in speech perception concerns whether top-down lexical information influences early perceptual representations. This was addressed using the event-related potential (ERP) technique to measure semantic priming effects on the auditory N1, an index of initial acoustic cue encoding. Participants saw visual primes that formed a clear association with the target (Association: "MARCHING *band*"), led to no specific association (Neutral: "BUTTER *bomb*"), or consisted of a non-word Mask. Auditory targets were stop consonants varying in voice onset time (VOT) between voiced (/b,d,g/) and voiceless (/p,t,k/) pairs. Participants were faster to identify the initial sound of the target in the Association condition than the Neutral and Mask conditions, and ERP responses showed the expected bottom-up effect of stimulus VOT (larger N1s for shorter VOTs). In Experiment 1, Association primes produced smaller N1s when targets were perfectly predictive, suggesting a top-down attentional effect. In Experiment 2, ambiguous and unexpected VOTs were added, and the results demonstrated that ambiguous VOTs in the Association condition were encoded similarly to the voicing endpoint that matched the semantic context (i.e., larger N1s for voiced expectations). These results provide the first ERP evidence that top-down lexical information directly influences early perceptual responses.

Invited Paper

9:40

3aPP6. Spoken word segmentation and contextual predictability. Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 1290, Eugene, OR 97403, mbaesebe@uoregon.edu) and Tuuli Morrill (none, Fairfax, VA)

During speech perception, listeners rely on context when parsing the speech signal. That is, a listener's interpretation of the speech signal does not only rely on bottom-up phonetic cues in the signal. Instead, listeners use a variety of cues to interpret the best parse of a speech signal. In the current study, we examine how spoken word segmentation of potentially ambiguous stretches of speech is impacted by a variety of contextually based acoustic-phonetic cues (e.g., speaking rate) and the interactions of these cues with contextually-based linguistic knowledge (e.g., collocation strength). We additionally ask how the strength of these predictive variables varies as a function of native speaker status. That is, do non-native speakers use contextual factors in the same way that native speakers do? Our results suggest that while listeners all use both acoustic-phonetic cues and linguistic knowledge when determining the most likely parse of speech, relianceon these cues varies as a function of language status. We will discuss the implications of these results for our understanding of speech perception, generally speaking.

Contributed Paper

10:00

3aPP7. Visual primes, speech intelligibility, and South Asian speech stereotypes. Veera S. Vasandani (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, vasan007@umn.edu), Molly E. Babel (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada), and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Visual primes that suggest social attributes about a talker can affect listeners' speech perception. Recent work by Babel and Russell [*J. Acoust. Soc. Am.* **137** (2015)] showed that speech intelligibility can decrease when listeners are shown a picture of the speaker when the talker is ethnically Chinese than when the talker is not. They reason that listeners associate Chinese faces with nonnative English accents, which hinders intelligibility. This finding has important implications for our understanding of real-world speech intelligibility when talkers' and listeners' race or ethnicity differ and when these are associated with different language backgrounds. Due to ongoing demographic shifts in the US and Canada, interactions between older and younger adults are often between people with different racial, ethnic, and linguistic backgrounds. This project is a part of a larger research collaboration aimed at understanding the extent and nature of effects of race and ethnicity on speech intelligibility across the lifespan and across levels of hearing acuity. In this talk, we will discuss the results of the first of these experiments, in which we examine speech intelligibility in noise using a large set of talkers whose voices are paired with faces of individuals who are White or South Asian.

10:15–10:30 Break

10:30

3aPP8. Variability in context effects on rate adaptation within individuals. Christopher C. Heffner and Emily B. Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., U-1085, Storrs, CT 06269, christopher.heffner@uconn.edu)

Rate adaptation is a crucial aspect of speech perception. Listeners can rapidly adapt to a wide variety of speech rates. Recent evidence has indicated that these effects can emerge even over the course of single sentences: studies of what are often termed "distal context effects" have shown that the rate context at the beginning of a sentence can affect the perception of words at the end of a sentence. These effects have been found across contrasts and languages. Yet the consistency of these effects within individuals has almost never been probed. Here, we report a series of experiments that examined whether distal context effects are consistent within individual talkers. We find surprisingly low test/retest reliability in distal context effects. Namely, the strength of the effects within individuals in one session or in one set of items had almost no bearing on the strength in a second session or second set of items. This was true even when the item set was highly constrained, to mitigate item effects in the stimuli. This suggests that a more nuanced understanding of individual variation in context rate effects is necessary.

10:50

3aPP9. Development of speech recognition in noise or two-talker speech: Context effects related to response alternatives and sentence meaning. Emily Buss (UNC Chapel Hill, 170 Manning Dr., G190 Physicians, Chapel Hill, NC 27599, ebuss@med.unc.edu) and Lori Leibold (Boys Town National Res. Hospital, Omaha, NE)

Speech recognition in normal-hearing adults is affected by pragmatic restrictions on the target content. For example, masked sentence recognition is better when target speech is composed of semantically meaningful compared to anomalous sentences. Similarly, masked word recognition is better when assessed in a close-set than an open-set task, even after accounting for changes in chance performance; this effect is most pronounced when the response alternatives in the closed-set task are acoustically distinct. In both the cases, restricting the set of plausible responses reduces the fidelity of acoustic cues necessary to perform the task. Although young school-age children benefit from this type of context, the magnitude of benefit relative to that observed for adults depends on the masker type. The benefit associated with increasingly restricted response alternatives is similar for young children and adults when the masker is noise, but young children derive little or no benefit when the masker is two-talker speech. Children and adults also differ with respect to the benefit of semantic context for sentences presented in noise or two-talker speech. Potential factors responsible for these developmental effects will be discussed, including maturation of auditory stream segregation, working memory, and acoustic/phonetic templates supporting word recognition.

11:10

3aPP10. Backwards and indirect context effects in accommodating gender differences in speech. Matthew Winn (Speech-Language-Hearing Sci., Univ. Of Minnesota, 1417 NE 42nd St., Seattle, WA 98105, mwinn83@gmail.com) and Ashley Moore (Geneva Foundation, Seattle, WA)

Listeners accommodate large amount of acoustic variability across talkers and phonetic contexts when categorizing speech sounds. This study examines accommodation of gender-related talker differences, which can occur in reverse (in sound sequence AB, sound B affects perception of sound A), suggesting more complicated mechanisms than peripheral auditory contrast enhancement alone. A continuum of fricatives "sh" and "s" was preappended to vowels whose acoustic parameters were manipulated by various properties ranging from typically feminine to masculine and vice versa. Other conditions provided matching or mis-matching visual cues. We examined the weighting of cues to talker gender, the duration of exposure to sound needed to obtain the full context effect, and the influence of hearing status, using listeners with cochlear implants (CIs). CI listeners used different cues to show a comparable gender context effect and also appear to use proxy cues for acoustic properties perceived more clearly by people with normal hearing (NH). All listeners showed implicit knowledge of how acoustic patterns are selectively reverse-compatible with different articulation sequences. NH listeners needed only 30 ms for full adjustment, while CI listeners' data were much more variable. Collectively, these results show a complex process of auditory, visual and gesture-aware adjustment to a talker's voice.

11:30

3aPP11. Diagnostic precision of open- versus closed-set word recognition. Tzu-Ling J. Yu and Robert S. Schlauch (Speech-Language-Hearing Sci., Univ. of Minnesota, Twin Cities, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, yuxxx583@umn.edu)

The precision of forced-choice (closed-set) and open-ended (open-set) word recognition (WR) tasks for identifying a change in hearing was examined. WR performance for closed-set (4 and 6 choices) and open-set tasks was obtained from 70 listeners with normal hearing. Speech recognition was degraded by presenting words from the Modified Rhyme Test in speech-shaped noise (-8, -4, 0, and 4 signal-to-noise ratios) or by processing words using a sinewave vocoder (2, 4, 6, and 8 channels). The results for the two degraded listening conditions yielded similarly shaped, monotonically increasing psychometric functions. The closed-set tasks had shallower slopes and higher scores than the open-set task for the same condition. Fitted, average psychometric functions were the input to a computer simulation conducted to assess the ability of each task to identify a change in hearing. Individual data were also analyzed using 95% confidence intervals for significant changes in scores for words and phonemes. These analyses found the following for the most to least efficient condition: open-set (phoneme), open-set (word), 6-choice closed-set, and 4-choice closed-set. Greater than an order of magnitude more trials were needed for the 4-choice condition to equal the precision of the open-set word condition scored by the percentage of correct phonemes.

Session 3aSAa

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

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

Contributed Papers

8:00

3aSAa1. Propagation and scattering of Lamb waves with infinite wavelength. Clemens Gruensteidl (Mech. Eng., Univ. of Colorado at Boulder, 1111 Eng. Dr., UCB 427, Boulder, CO 80309, clgr8334@colorado.edu), David Stobbe (Lawrence Livermore National Lab., Boulder, CO), and Todd W. Murray (Mech. Eng., Univ. of Colorado at Boulder, Boulder, CO)

The dispersion characteristics of Lamb waves depend on the thickness and the elastic properties of a plate. At the cut-off frequencies (when the wavenumber goes to zero), Lamb waves are standing waves associated with plane longitudinal or shear waves reflecting up and down between the surfaces of the plate at normal incidence. In the general case, these thickness resonances have an infinite wavelength and do not carry energy along the plate. However, if the material properties are selected such that longitudinal and shear thickness mode resonances of the same symmetry coincide, then the elastic fields associated with the resonances couple to produce a wave with an infinite wavelength that propagates energy along the plate surface. Intriguing effects associated with this phenomenon can be observed in aluminum alloy plates where near-degeneracy between resonances occurs. We show that waves generated near the cutoff frequency in such cases spread from the excitation point and produce a spatially uniform oscillation over the plate surface. These infinite wavelength waves show angle independent mode conversion upon encountering the free edge of a plate or thickness step on the surface. Furthermore, we demonstrate that these peculiar waves flow around plate defects and are remarkably insensitive to scattering.

8:15

3aSAa2. Calculating the full leaky Lamb wave dispersion characteristics. Daniel A. Kiefer, Michael Ponschab, and Stefan J. Rupitsch (Chair of Sensor Technol., Univ. of Erlangen-Nuremberg, Paul-Gordan-St. 3/5, Erlangen 91052, Germany, daniel.kiefer@fau.de)

Lamb waves are mechanical waves guided in a plate with stress-free boundaries. In practice, the plate is often in contact with a fluid. The propagation of guided waves may change substantially in this case due to interaction with the adjacent medium, e.g., acoustic radiation may lead to leakage of energy into the fluid domain. The resulting plate modes are called leaky Lamb waves, and their attenuation due to leakage is an important parameter for the design of ultrasonic devices that exploit such waves. Modern methods to calculate dispersion characteristics of guided waves, i.e., the wavenumber-frequency relationship, solve the corresponding discretized eigenvalue problem. This kind of model for leaky Lamb waves with the analytically exact fluid interaction leads to a nonlinear eigenvalue problem, which is difficult to solve. We present a change in variables that yields an equivalent polynomial eigenvalue problem. The latter can be rewritten as a linear eigenvalue problem in state space. As a result, conventional numerical eigenvalue solvers can be used to robustly determine the sought wavenumbers. In contrast to traditional root-finding of the characteristic equations, the presented method guarantees to find all solutions and naturally handles complex wavenumbers. Additionally, no initial guesses for the wavenumbers are required.

8:30

3aSAa3. Temporally weighting a time varying noise field to improve Green function retrieval. Richard Weaver and John Yoritomo (Dept. of Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL, r-weaver@uiuc.edu)

We consider the retrieval of Green functions G from the correlations of non-stationary non-fully diffuse noise incident on an array of sensors. Multiple schemes are proposed for optimizing the time-varying weights with which correlations may be stacked. Using noise records created by direct numerical simulation of waves in a two-dimensional multiply scattering medium, cases are shown in which conventional stacking does a poor job and for which the proposed schemes substantially improve the recovered G, rendering it more causal and/or more symmetric, and more similar to the actual G. It is found that the schemes choose weights such that the effective noise field is close to fully diffuse, with little direction dependence to its intensity.

8:45

3aSAa4. A uniform theory of diffraction formulation for acoustic radiation modes. Caleb B. Goates, Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com), and Jonathan Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Acoustic radiation modes have recently regained the attention of the literature, with applications ranging from active structural acoustic control to acoustical holography. Typically the calculation of these modes requires boundary element methods for all but flat plate and full sphere geometries. This paper presents a radiation mode formulation for cylindrically curved structures with small curvature, based on the uniform theory of diffraction. This formulation shows promise, as it produces analytical expressions for the radiation resistance matrix, leading to computationally efficient radiation mode calculation. Radiation modes are shown for cylindrically curved plates, and the calculation of sound power with these modes is compared to that calculated using boundary element methods. The extension of this formulation to other curved structures is discussed.

9:00

3aSAa5. Extreme value statistics in flow-induced vibration over long time intervals. Connor McCluskey, Stephen Conlon, and Manton Guers (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804-0030, cxm1198@arl.psu.edu)

Cyclical loading on structures can lead to fatigue damage, and damage accumulation can be worsened from peak loading. In order to predict these peaks, extreme value statistics can be applied to limited vibration data. Tests must be conducted for long enough durations so that a representative population can be obtained. In extreme value statistics, the Generalized Extreme Value (GEV) distribution's parameters are also dependent on measuring a representative sample. As it is often uneconomical to test for long periods of time, the proposed experiment aims to look at the behavior of extreme value

statistics over long time records to determine if a "minimum requirement" is possible. A cantilever hydrofoil is vibrated under three flow conditions at two angles of attack. Extreme value statistics are applied to compare parameters and distributions for different record length increments. Statistics of each increment are used to generate return plots for the prediction of repeated tests. Errors are quantified to determine the accuracy of the different record lengths. The results will indicate how testing length influences GEV parameters and prediction in vibration, giving insight into duration requirements for future fatigue tests.

9:15

3aSAa6. Case study on the use of Comsol Multiphysics for undergraduate aeroacoustic research. Christopher Jasinski (Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06110, chrismjasinski@gmail.com)

The aerodynamic drag imposed by acoustic liners for commercial aircraft is a growing area of research. With the next generation of aircraft being designed for quietness and fuel-efficiency, this is expected to be a continued hot topic for the foreseeable future. There is substantial motivation to use computational techniques to evaluate the acoustic liner system, but the finite element analysis tools necessary are often inaccessible to undergraduates. Many of the mechanical engineering students at the University of Hartford learn the basics of COMSOL Multiphysics simulation software through their coursework in fluid mechanics, and this study aims to leverage this skill into more advanced research. As part of a senior capstone project, a pair of University of Hartford undergraduates will be using COMSOL to determine the acoustic impedance for various liner types and assess methods of determining the skin friction coefficient of the liner. These results will be compared to experimental work conducted over the last few years at the University of Notre Dame and NASA Langley Research Center. Experimental results, including direct drag and more detailed flow measurements, will be presented along with the computational results from COMSOL and a reflection on the ease of use for undergraduate researchers.

9:30

3aSAa7. Using measured data to enhance and extend vibro-acoustic performance predictions. Arnau Clot, Robin S. Langley (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, United Kingdom, ac2107@cam.ac.uk), Joshua Meggitt (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), David Hawes (Dept. of Eng., Univ. of Cambridge, Cambridge, United Kingdom), Andy S. Elliott, and Andy Moorhouse (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom)

Modern industries usually need to ensure that their manufactured products meet certain vibro-acoustic requirements. Therefore, they have a clear need for models that can predict the broadband dynamic response of structural components at the design stage. The use of a hybrid deterministic-statistical formulation has been shown to be a suitable solution for predicting the response of a complex system in the mid-frequency range. This work explores two potential uses of experimental data to enhance and extend the applicability of the hybrid deterministic-statistical approach. Measured data are used to represent, first, complex vibration sources and, second, junctions between different structural components. The proposed uses are tested in a case study consisting of a deterministic source structure coupled to a statistical plate receiver. The approach is validated by comparing the predicted vibration response of the receiver plate to the one obtained by experimentally randomising the plate. The results show that a good agreement is obtained, both for the statistics of the receiver response and for the dynamic properties related to a point junction. It is concluded that the use of measured data can clearly extend the applicability hybrid models.

9:45–10:00 Break

10:00

3aSAa8. A high-frequency model of a circular beam with a T-shaped cross section. Andrew J. Hull and Daniel Perez (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil)

This talk derives an analytical model of a curved beam with a T-shaped cross section for use in the high frequency range, defined here as approximately 1 kHz to 50 kHz. The T-shaped cross section is composed of an outer web and an inner flange. The web in-plane motion is modeled with two-dimensional elasticity equations of motion, and the left portion and right portion of the flange are modeled separately with Timoshenko shell equations. The differential equations are solved with unknown wave propagation coefficients multiplied by Bessel and exponential spatial domain functions. These are inserted into constraint and equilibrium equations at the intersection of the web and flange and into boundary conditions at the edges of the system which produces 24 linear algebraic equations. These are solved to yield the wave propagation coefficients and this gives a corresponding solution to the displacement field in the radial and tangential directions. An example problem is formulated and compared to solutions from fully elastic finite element modeling. The accurate frequency range of this new model compares very favorably to finite element analysis up to 50 kHz. This new analytical model is over 3 magnitudes faster than the corresponding finite element models.

10:15

3aSAa9. Nonlocal homogenisation scheme for lossless and anisotropic metafluids and phononic crystals. Navid Nemati, Johann Guilleminot (Civil and Environ. Eng., Duke Univ., Hudson Hall, Office 167, Durham, NC 27708, navid.nemati@duke.edu), and Mário G. Silveirinha (Instituto Superior Técnico, Univ. of Lisbon, Lisbon, Portugal)

We present a systematic homogenisation approach to describe macroscopic dynamics of generic periodic acoustic materials composed of a rigid structure filled with a lossless fluid. This approach takes temporal dispersion as well as spatial dispersion effects into account, and can characterise, in a general manner, the oblique propagation and anisotropic effects. The theory is formulated such that all effects are encoded in, only, the effective frequency and wavevector dependent density, which can be calculated through a source-driven problem. Also, we demonstrate that the theory can homogenise materials even in frequency band gaps, including those that correspond to local-resonance phenomena or to Bragg scattering.

10:30

3aSAa10. Far-field acoustic subwavelength imaging and edge detection. Chu Ma and Nicholas X. Fang (Dept. of Mech. Eng., Massachusetts Inst. of Technol., MIT Bldg. 3-466, 77 Massachusetts Ave., Cambridge, MA 02139, machu@mit.edu)

The resolution of far-field acoustic imaging is limited due to the loss of the evanescent field that carries subwavelength information. In this work, we report on the design and experimental realization of an acoustic far-field subwavelength imaging system based on a transmitting/receiving pair for wave vector filtering and conversion. In the near-field, the transmitter is composed of a resonator array for pre-filtering the evanescent wave, and an acoustic phase grating for converting the filtered evanescent wave into propagating wave. In the far-field, a receiver that is spatially symmetrical to the transmitter is used to convert the propagating wave back to evanescent wave and perform post-filtering. By tuning the resonance frequency of the resonator array and period of the phase grating, different spatial frequency bands can be separated and projected to the far-field. Far-field imaging and edge detection of subwavelength objects are experimentally demonstrated. The proposed system brings new possibilities for far-field subwavelength wave manipulation, which can be further applied to medical imaging, nondestructive testing, and acoustic communication.

10:45

3aSAa11. Data-driven approaches for damage-type classification in vibrating square plates. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

Acoustic radiation from a mechanical structure due to broadband forcing is inherently dependent on the structure's material, geometry, boundary conditions, and mechanical status. Measurements of this acoustic or vibrational response can be used to detect mechanical changes (i.e., damage) when compared to known baseline measurements of the structure. Often, however, knowledge of the *type* of damage is useful for subsequent considerations. Even for relatively simple structures, like flat plates, the lack of simple

solutions to problems involving localized damage makes analytical treatments challenging. This is further complicated since the location and severity of damage are typically unknown a priori. We present a data-driven approach for classification of various forms of damage—including cuts, localized mass changes, delamination, and boundary changes—in a vibrating clamped square plate. Frequency response curves are generated using FEA of a 30-by-30-by-0.3-cm aluminum plate with various types and severity of damage. Features (including changes in peak-locations, -amplitudes, and -widths compared to baseline) are extracted and used to train classifiers, with performance quantified using auxiliary test data. Comparisons are made between classifier types, including nearest neighbor methods, discriminant analysis, and support vector machines. [Work supported by NAVSEA through the NEEC and the US DoD through an NDSEG Fellowship].

Session 3aSAb

Structural Acoustics and Vibration, Engineering Acoustics, and Noise: Noise and Vibration in Rotating Machinery

Robert M. Koch, Cochair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Elizabeth A. Magliula, Cochair
Division Newport, Naval Undersea Warfare Center, 1176 Howell Street, Bldg 1302, Newport, RI 02841

Chair's Introduction—10:45

Invited Papers

10:50

3aSAb1. Decreasing the radiated acoustic and vibration noise of autonomous underwater vehicles. Gerald L. D'Spain, Richard Zimmerman, Dennis Rimington, and Scott A. Jenkins (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

An outstanding challenge in ocean engineering is decreasing the radiated acoustic and vibration noise of autonomous underwater vehicles (AUVs) while transiting at high speed. At low speed below 4–5 kt, our previously published results for both a modified mid-size, prop-driven AUV and a large (30 liter) buoyancy driven glider demonstrate that, with proper engineering, own-platform-created noise levels recorded by hull-mounted hydrophone arrays can be decreased below typical background ocean noise levels across the frequency band above a few hundred hertz. Below this frequency, the remaining noise is primarily vibration induced. Recent efforts have focused on decreasing propulsion noise at high speed from prop-driven platforms. Since buoyancy can be changed only over a very short portion of the dive cycle, buoyancy-driven propulsion systems provide an alternative approach to achieving low-self noise, high-speed, underwater flight. However, vibration and flow noise created by hydrodynamic forces still can present challenges at low frequencies. [Work supported by the Office of Naval Research and BP.]

11:10

3aSAb2. Perception of complex tones in noise. Jan Hots, Gloria-Tabea Badel, and Jesko L. Verhey (Dept. of Experimental Audiol., Otto von Guericke Univ. Magdeburg, Leipziger Straße 44, Magdeburg 39120, Germany, jan.hots@med.ovgu.de)

We are often surrounded by technical sounds that contain tonal components, which usually are generated by rotating parts of machinery. Tonal components commonly reduce the pleasantness of a sound, i.e., these sounds are perceived as less pleasant compared to equal-level sounds without tonal components. To consider the influence of tonal components on noise pollution, different standards include sections dedicated to this effect. For single tonal components, previous data showed that tone adjustments calculated on the basis of the German standard 45681 account quite well for the reduced pleasantness. However, the standard is limited to single tonal components only. It was also shown that the partial loudness of the tonal portion corresponds to the perceived magnitude of tonal content (also known as tonalness or tonality). The present study investigated the perception of complex tones in noise. To this end, the loudness of the tonal portion of the sound is determined for sounds containing a single tone or a complex tone with two or four components at different levels above the masked threshold. It is shown that an increase in components results in an increase in the magnitude of the tonal content. This likely also affects the pleasantness of the sound.

Contributed Papers

11:30

3aSAb3. Quadcopter propeller noise and its control. Timothy A. Brungart, Stephen T. Olson, Brian L. Kline, and Zachary W. Yoas (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804-0030, tab7@arl.psu.edu)

Simple scaling analysis is used to show that truly significant reductions in propeller noise are possibly by increasing the propeller diameter while reducing its rotational speed, thereby reducing its blade tip speed, while maintaining a given level of static thrust. The scaling analysis is also used to show that efficiency improvements accompany the reductions in radiated noise. The significant reductions in noise and the increase in efficiency predicted were verified experimentally by measuring the radiated noise and power requirements for both large, slowly rotating propellers and small, high speed propellers, at equivalent static thrusts, where the large propellers serve as possible replacements for the small propellers in typical quadcopter applications. The bulk of the noise reductions offered by large and slowly rotating propellers compared with small, high speed propellers is shown to be maintained at equivalent net force (thrust minus weight conditions), as required for practical implementation into a quadcopter, even using readily available hobby grade components. Further significant noise reductions are possible with lightweight and custom engineered components.

11:45

3aSAb4. Nonlinear response of a flanged fastened joint due to acoustical sine sweep excitation. Trevor W. Jerome, Micah R. Shepherd, and Stephen Hambric (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, MS 3220B, State College, PA 16804-0030, twjerome@gmail.com)

Fastened joints can significantly complicate the prediction of damping and resonance frequencies of built-up systems. This complexity often leads to over-designing for safety, which drives up manufacturing and operational costs. To improve understanding of the dynamic response of fastened joints, impact hammer and acoustically driven excitation of fastened plates at a flange have been measured. These findings add experimental data that are not well represented in the literature for such joints. The structure was excited acoustically, and acceleration was measured at multiple locations on the flange and plate. Backbone curves were generated with sinusoidal sweeps of acoustic input energy. These curves highlight the nonlinearity of the system, which varies according to fastener preload. Generally, as preload torque values decrease, nonlinearity increases with shifts observed in damping, stiffness, and resonance frequency. Data presented will be used to create a model for better predicting the global dynamic properties of built-up structures.

WEDNESDAY MORNING, 15 MAY 2019 GRAND BALLROOM C, 9:00 A.M. TO 12:00 NOON

Session 3aSC

Speech Communication: Developmental and Clinical Populations (Poster Session)

Terrin N. Tamati, Chair
Department of Otorhinolaryngology / Head and Neck Surgery, University Medical Center Groningen, Hanzeplein 1, Groningen 9700RB, The Netherlands

All posters will be on display, and all contributors will be at their posters from 9:00 a.m. to 12:00 noon. To give contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

3aSC1. A real-time, automated tongue tracking method for ultrasound biofeedback in speech therapy. Jack A. Masterson, Sarah R. Li, Hannah M. Woeste (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, ML 0586, Cincinnati, OH 45267-0586, masterjk@mail.uc.edu), Sarah Dugan (Commun. Sci. and Disord., Univ. of Cincinnati, Dayton, OH), Neeraja Mahalingam (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Suzanne E. Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Michael A. Riley (Psych., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Midsagittal ultrasound images of the tongue surface are currently used as feedback on tongue motion for speech therapy, but can be difficult to interpret. Ultrasound biofeedback would benefit from a simplified, immediate representation of tongue motion. Tongue tracking in ultrasound is complicated by reflections from structures near the tongue surface, complex tongue shapes, and rapid, extreme tongue movements. Available programs that trace contours of the tongue surface have not been implemented in real time and can require extensive user input. We offer a tongue tracking method that can operate in real time and requires minimal user input. This method applies a low-pass filter to the B-mode image and then maps the tongue surface based on local brightness maxima within search regions estimated by 2nd-order Taylor series expansions in space and time. Required user input includes a graphically selected calibration point near the tongue surface and relative brightness thresholds for anterior and posterior ends of the tracked surface. Preliminary results show that this method can capture complex, rapid movements of tongue parts during production of American English /r/. Efficient tongue tracking enables real-time analysis of tongue dynamics, offering a valuable tool for speech therapy.

3aSC2. An ultrasound study of American English laterals produced by first graders. Sherman D. Charles and Steven M. Lulich (Speech & Hearing Sci. and Linguist, Indiana Univ., 1610 S Dorchester Dr. Apt. 49, Bloomington, IN 47401, sdcharle@indiana.edu)

Longitudinal studies of child speech production development have traditionally relied on perceptual evaluation and acoustics to track acquisition norms and phonological patterns. The present study administers novel techniques in speech imaging to investigate the articulation and acoustics of speech sounds in developing children, focusing on the lateral sound /l/ as produced by first graders. Real-time three-dimensional ultrasound images with synchronous audio and video recordings were collected from L1 English speaking children. The stimuli were sourced from the GFTA-3—a clinical picture-naming instrument—which was administered by a speech-language pathology student-clinician. Three-dimensional tongue surface reconstructions and the accompanying audio signals were then compared across tokens and across speakers to find common and divergent patterns for lateral sounds. The results show highly variable articulatory strategies that nevertheless can be categorized into three groups. The wide articulatory variety is not obviously tied to acoustic variability, which is relatively small; formant positions are relatively consistent across tokens and across subjects, with few outliers. From the acoustics alone, these children consistently produce what is traditionally described as the dark /l/, but articulation suggests greater variety.

3aSC3. Dyslexia impedes the processing of vocoded speech. Robert A. Fox, Ewa Jacewicz, Gayle Long, and Zane Smith (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

Research has shown that phonological impairment in dyslexia (DYS) is associated with a deficit in recognizing fine-grained details in voices of multiple talkers in both clear and degraded speech [Fox *et al*., *J. Acous. Soc. Am.* **141**, 3839 (2017)]. These results suggest that DYS listeners may have difficulties in the processing of vocoded speech due to the absence of harmonic information. Given the extensive variability in speech including gender and dialect of the speakers, we can predict that DYS individuals may have limitations using cochlear implants. In the current study, 20 DYS adults and 20 corresponding control listeners heard a set of short sentences/phrases produced by male and female talkers from Central Ohio or Western North Carolina. The utterances were unprocessed or processed through a noise-source vocoder at 4, 8, 12, and 16 channels. In separate experiments, listeners were asked to identify the dialect of the speaker (an ID task) or to repeat the sentence/phrase heard (an intelligibility task). The ID responses (A' scores) of the DYS listeners were lower than those of the controls at every stimulus level. Similarly, intelligibility responses demonstrated that DYS listeners performed more poorly than did control listeners at every stimulus level.

3aSC4. Implant location in type 1 thyroplasty for unilateral vocal fold paralysis: Measurements from excised canine larynges. Alexandra Maddox, Charles Farbos de Luzan, Liran Oren, Sid M. Khosla, and Ephraim Gutmark (Univ. of Cincinnati, 1118 Broadway St., Cincinnati, OH 45202, maddoxa219@gmail.com)

Patients with unilateral vocal fold paralysis (UVFP) complain of a soft breathy voice that is hard to understand. Thyroplasty Type 1 (TT1), the most common surgical intervention for UVFP, uses an implant to push over the membranous fold. There remains controversy regarding TT1 procedures. Questions include the optimal shape and size for the implant, where to place the implant and whether an arytenoid adduction should be added. Despite several studies, no optimal technique has been found, and the revision rate has reported to be as high as 12%–25%. This study aims to understand the impact of implant location on vocal fold vibration. TT1 procedures were performed on excised canine larynges using silastica implants placed either at the glottis or just below the glottis to medialize the folds. Larynges were then made to phonate at various subglottal pressures (Psg), and measurements were taken of the acoustics, supplied flow rate, and Psg. From these measurements, the glottal efficiency was calculated, and the acoustic signal was analyzed to determine the quality of the sound source. On average, larnges medialized in the subglottis had higher glottal efficiency and cepstral peak prominence than those medialized at the glottis, indicating that lower implant location is preferable.

3aSC5. Dysarthric speech perception: Comparison of training effects on human listeners versus automatic speech recognition tools. Michael F. Lally (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 N Wright St., Urbana, IL 61801, mlally2@illinois.edu), Heejin Kim (Linguist, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Lori A. Moon (Psych., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Dysarthric speech, a motor speech disorder associated with neuro-motor conditions, poses a challenge for automatic speech recognition (ASR) due to its acoustic characteristics and the comparatively limited volume of available data. This study investigates ASR toolkits' recognition abilities on speech from speakers affected with cerebral palsy (CP) at different intelligibility levels, by comparing them against human listeners' performance. We ask: (1) how intelligible is speech from CP-affected speakers to ASR toolkits trained on non-dysarthric speech? (2) how does this performance compare to naïve human listeners? and (3) as familiarized human listeners better understand dysarthric speech, to what degree is dysarthric speech more intelligible to similarly familiarized ASR toolkits? Using the UA-Speech Database (Kim *et al*., 2008), we test the ASR system with two training methods: strong supervision, with both audio and orthography feedback in training, and unsupervised methods, with only audio signals, following Kim and Nanney's (2014) experiments with human listeners. ASR accuracy is measured by the word error rate in word transcription tests. Findings reveal the extent to which supervision affects ASR models in comparison to human listeners. Implications regarding how to improve the adaptation techniques to dysarthric speech for both ASR and human listeners are presented.

3aSC6. Mandarin Chinese consonant production of patients with cleft palate. Heng Yin, Xi Wang, Chunli Guo, Jingtao LI, Yang Li, and Bing Shi (Dept. of Cleft Lip and Palate Surgery, West China College of Stomatology, West China Hospital of Stomatology, Sichuan Univ., Chengdu, Sichuan 610041, China, yinheng@scu.edu.cn)

Eighteen Mandarin Chinese consonants were recorded from 259 patients with cleft palate in a /CV/ context. The age of participants ranged from 8 to 40 years old. Participants had five types of cleft palate: submucous cleft palate, hard and soft cleft palate, unilateral complete cleft palate, bilateral complete cleft palate, and congenital velopharyngeal insufficiency. All consonants were transcribed by speech-language pathologists. Results showed that the accuracy of consonant production was significantly affected by the type of cleft palate, consonant category, and their interaction. The correctness of consonant production ranged from 45.8% for the type IV cleft palate (e.g., BCCP) to 71.5% for Submucous cleft palate. In particular, for the manner of articulation, nasals had significantly higher accuracy than stops, fricatives, and affricates, while for the place of articulation, palatal and labial sounds were problematic with low accuracy. In addition, the errors of consonant production, such as replacement, compensation, and omission, were also dependent on the type of cleft palate and consonant category and their interaction.

3aSC7. Efficacy of two speech-perception training methods for aided listeners with mild-to-moderately-severe age-related hearing losses. James D. Miller, Charles S. Watson, and Gary R. Kidd (Res., Commun. Disord. Technol., Inc., 3100 John Hinkle Pl, Ste 107, Bloomington, IN 47408, jamdmill@indiana.edu)

Preliminary analyses of interrelations among variables that correlate with measures of speech perception by aided listeners with mild-to-moderately-severe age-related hearing loses are described. The SPATS Group of nearly 120 hearing-aid users was trained in quiet and noise to identify syllable-constituents and to identify words in simple sentences. The Listen Group of about 60 aided listeners listened for an equal amount of time to recorded narratives. All were given audiometric, working memory, intelligence tests, a battery of psycho-acoustic tests, and a battery of speech-perception tests prior to training. The audiometric and speech perception tests were repeated after training and again after a 3-month retention period. Prior to training, there were large individual differences speech perception scores

that could only be partially accounted for by the severity of audiometric loss. Training appears to be more effective for the SPATS Group than for the Listen Group. While improvements with training were generally modest, those who were very good or very poor initially tended to show little or no improvement, while those with middling scores tended to show more improvement. Sentence scores are highly correlated with syllable-constituent scores and with the use of context, and the use of context is correlated with working memory. (Miller and Watson are stockholders in communication Disorders Technology, Inc., and may profit from sales of some the software used in this study.)

3aSC8. Contributions to diminished perceptual learning in individuals with language impairment. Nikole Giovannone and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT, nikole.giovannone@uconn.edu)

The lexical context in which sounds occur can help listeners mitigate the challenges of the lack of invariance in the speech signal. Through lexically guided perceptual learning, phonetic category structure can be dynamically modified given repeated exposure to potentially ambiguous sounds embedded within familiar lexical contexts. Individuals with language impairment (LI) show deficits in general auditory processing and speech perception, which may influence the development of stable phonetic category structure. However, it is not yet known whether deficits in phonetic category structure in individuals with LI are related to impairment in the learning mechanisms that guide adaptation or in feedback links from the lexicon to speech sounds. To assess these sources of deficit, participants will complete measures of receptive language ability, a task assessing lexical recruitment (i.e., the Ganong effect), and a lexically guided perceptual learning task. If impairment in phonetic category structure stems from weaker top-down lexical influences, then performance on the lexical recruitment task should be (1) diminished in individuals with LI and (2) predictive of lexically guided perceptual learning. If diminished adaptation instead reflects a disrupted learning mechanism, then individuals with LI and typical participants should perform comparably on the lexical recruitment task despite showing differences in lexically guided perceptual learning.

3aSC9. The effect of talker variability on word recognition under cochlear implant simulation. Terrin N. Tamati and Deniz Baskent (Dept. of Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Hanzeplein 1, Groningen 9700RB, The Netherlands, t.n.tamati@umcg.nl)

Normal-hearing (NH) listeners have been shown to be less accurate and/or slower to recognize spoken words when the talker changes from trial to trial, compared to when the talker remains the same. Less is known about the effect of talker variability on cochlear implant (CI) users, who display a limitation in talker perception. As a first step, the current study investigated how limited talker information via CI simulation influences word recognition. In two experiments, NH listeners completed a word naming task in different talker conditions (single (ST), multiple female (MT-F), and multiple female/male (MT-M)) and simulation conditions (12, 8, and 4 vocoder channels). An effect of talker variability was observed, but only when talker differences were maximized. Listeners were less accurate, but not slower, at recognizing words under the MT-M condition, compared to ST and MT-F conditions. Talker variability effects did not vary by the simulation condition. These results are consistent with previous studies with NH listeners, showing similar performance in conditions where talker changes were not detected but talker differences were large. Taken together, these findings suggest that limitations in talker perception for NH listeners under CI simulation, and potentially CI users, alters the perceptual strategy for recognizing words under multiple-talker conditions.

3aSC10. Flow characteristics as a function of velopharyngeal gap size. Michael Rollins (Biomedical Eng., Univ. of Cincinnati, MSB 6303, 231 Albert Sabin Way, Cincinnati, OH 45267, rollinmk@mail.uc.edu) and Liran Oren (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH)

The classic paradigm for treatment of velopharyngeal insufficiency (VPI) is based on the assumption that the severity of the speech disorder correlates with the size of the velopharyngeal (VP) opening. However, several studies have documented that even small VP openings can cause significant speech distortion. To investigate the potential contribution of aeroacoustic sources to speech distortion in VPI-affected speech, we set out to determine how airflow characteristics change as a function of the VP gap size. To do so, a head and neck CT scan of a VPI patient sustaining the phoneme /z/ was obtained. From the CT scan, patient-specific phantoms of the upper airway were made extending from the glottis to the nasal and oral exits. Four phantoms were made: one with the original VP gap size and three with VP gap sizes artificially scaled by factors of 1.25, 1.5, and 2. Particle image velocimetry (PIV) was used to measure the velocity and turbulence fields along planar slices in each phantom. Average velocity fields for each VP gap size are presented, and the change in flow characteristics as a function of the VP gap size is discussed, along with the implications for aeroacoustic sources.

3aSC11. Detecting changes in lingual differentiation over the course of speech sound intervention. Heather M. Kabakoff, Graham T. Feeny (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, heather.kabakoff@nyu.edu), Mark Tiede (Haskins Labs., New Haven, CT), Jonathan L. Preston (Commun. Sci. and Disord., Syracuse Univ., Syracuse, NY), and Tara McAllister (Communicative Sci. and Disord., New York Univ., New York, NY)

"Lingual differentiation" describes the ability to articulate different regions of the tongue semi-independently. Children with speech sound disorder (SSD) exhibit fewer differentiated gestures than typically developing children (Gibbon, 1999) and show a greater degree of lingual differentiation after treatment than before treatment (Preston *et al*., 2018). In the current study, four preschool-aged children (4;1-5;4) were treated for error patterns identified on the HAPP-3 measure in 18 sessions of cycles treatment (Hodson and Paden, 1983) over six weeks. Ultrasound and audio recordings were collected as children named pictures containing a variety of target sounds at pre-treatment, during six within-treatment sessions, and at post-treatment. For each production, lingual contours were extracted from midsagittal ultrasound images and quantified using two complexity metrics: modified curvature index (Dawson *et al*., 2016) and number of inflections (Preston *et al*., 2018). Trained listeners transcribed each production and calculated the percentage consonants correct (PCC) at each time point using Phon speech analysis software (Hedlund and Rose, 2016). We ask whether incremental changes in lingual differentiation are detectable in preschool-aged children enrolled in treatment, and whether these changes correlate with the increase in production accuracy (PCC). We also examine gradient differences in lingual complexity for rhotic tokens transcribed as misarticulated.

3aSC12. Effect of number of channels and speech coding strategy on speech recognition in mid-scala electrode recipients. Katelyn A. Berg (Dept. of Speech & Hearing, Vanderbilt Univ., Ste. 8310, 1215 21st Ave. South, Nashville, TN 37232, katelyn.a.berg@vanderbilt.edu), Jack H. Noble, Benoit Dawant (Dept. of Elec. Eng. & Comput. Sci., Vanderbilt Univ., Nashville, TN), Robert Dwyer (Dept. of Speech & Hearing, Vanderbilt Univ., Nashville, TN), Robert Labadie (Dept. of Otolaryngol., Vanderbilt Univ., Nashville, TN), and René H. Gifford (Dept. of Speech & Hearing, Vanderbilt Univ., Nashville, TN)

This study investigated the number of channels needed for maximum speech understanding and sound quality in 5 (anticipated n = 10) adult cochlear implant (CI) recipients with mid-scala electrode arrays completely within scala tympani (ST). CI programs with 4–16 active electrodes using a CIS-based strategy were created along with two n-of-m programs (8-of-16 and 12-of-16). Measures of speech understanding and sound quality were assessed. Our hypothesis was that individuals with precurved electrodes localized in ST would have greater channel independence than previous studies demonstrating gains in performance beyond 8 channels. Participants demonstrated performance gains up to 8–10 electrodes for speech recognition and sound quality ratings. Significantly poorer performance was observed using an n-of-m strategy as compared to CIS conditions with 8+ electrodes. These data are in contrast with recent studies for precurved arrays (e.g., Croghan *et al*., 2017; Berg *et al*., in review) reporting significant improvement up to 16 to 22 channels. However, the current results are

consistent with previous literature (e.g., Fishman *et al*., 1997; Friesen *et al*., 2001; Shannon *et al*., 2011), demonstrating no more than 8–10 independent channels for CI recipients with straight arrays. Implications for device selection and the impact of electrode-to-modiolus distance will be discussed.

3aSC13. Acoustic measures of paraspeech in speakers with multiple sclerosis: A study of neuropsychological status and its role in diadochokinesis. Lynda Feenaughty (Univ. of Memphis, 4055 N Park Loop, Memphis, TN 38152, lynda.feenaughty@memphis.edu)

Paraspeech tasks are frequently incorporated in motor speech disorder assessments and include syllable diadochokinesis (DDK), the rapid repetition of alternating movements (Kent, 2015). DDK measures indicate movements of the oral articulators and may detect problems before specific functions such as speech are affected in neurodegenerative diseases such as multiple sclerosis (MS). Although there is growing appreciation from studies of various clinical populations that cognitive and speech motor processes influence or interact with one another (Kent, 2004), the impact of cognitive limitations of speakers with dysarthria on DDK performance is not well understood. Toward this end, the current study explored cognitive status as a factor in DDK performance for 48 speakers with MS and 12 healthy adults. All speakers were recorded as they repeated various syllables as quickly and steadily as possible. Standard acoustic criteria were used to obtain global and segmental temporal measures of DDK using speech analysis software. In addition, all speakers underwent rigorous cognitive testing (e.g., information processing efficiency). Because controversy continues over the use of DDK, this study sought to provide evidence for the clinical value of paraspeech in the assessment of motor speech disorders as well as cognitive status. [Work supported by the ASH Foundation and the University at Buffalo MDRF.]

3aSC14. Masking effects of perceptual cues on hearing-impaired ears. Siyan Guo, Cliston Cole, and Jont Allen (ECE, Univ. of Illinois, Urbana-Champaign, 306 N Wright St., Urbana, IL 61801, sguo16@illinois.edu)

The most common complaint from hearing-impaired (HI) subjects is I can hear the speech but I can't understand it. This has led us to investigate the strategy of the HI ear in understanding speech. In this study, the effects of the masking of primary cues in HI ears are analyzed, with and without the presence of conflicting cues. Two consonant-vowel (CV) identification experiments were conducted on 5 normal-hearing (NH) and 10 HI subjects. 4 plosive consonants /t,k,d,g/ paired with vowel /a/ were used as target stimuli. The CVs were presented at SNRs of 0, 9, and 18 dB. In experiment I, the primary cue for each CV was modified in 3 ways: removed, attenuated, and amplified. Experiment II was similar except the conflicting cues were removed. The results for both NH and HI groups show a large increase in probability of error in all SNR levels when the primary cue was removed. The results for Exp II display more variability. The HI subjects were divided into three groups based on their error: low error group (LEG), medium error group (MEG) and high error group (HEG). The analysis shows that HI listeners are using the same primary cue as NH ears. The strength of the primary cue is a critical quality for correct speech perception, especially in the presence of noise. The subjects in MEG and HEG exhibit sensitivity to the presence of conflicting cues. The removal of the conflicting cues demonstrates that HI ears use not only the primary cue but also conflicting cues, for correct speech perception.

3aSC15. Whole word scoring versus phoneme error scoring for audiological word recognition testing. Edward L. Goshorn, Jennifer Goshorn, and Jennifer Arnoult (Speech and Hearing Sci., PsychoAcoust. Res. Lab., Univ. of Southern Mississippi, 118 College Dr. #5092, Hattiesburg, MS 39401, edward.goshorn@usm.edu)

Conventional audiological word recognition testing uses whole word scoring for errant responses resulting in a percent correct score. This approach is found to be lacking in diagnostic utility. Recently, phoneme error scoring was suggested as a supplement to whole word scoring in attempt to improve diagnostic utility but is also limited in that it is highly correlated with the whole word score. This project examined the use of phoneme error scoring that applies an exponential equation to the type and the number of phoneme errors yielding a score that may be compared to absolute hearing sensitivity loss as measured by conventional audiograms. Because the number of words used in audiological word recognition varies, a correction for the number of stimuli is applied. Archival data, case study data, and simulated cases were used to evaluate and compare the diagnostic utility of three scoring approaches for errant responses: whole word, total phoneme errors, and an exponential phoneme error score (EPES) that weights consonant and vowel errors differently: EPES = number of phoneme errors times 2^{number} of consonant errors + 10^{number} of vowel errors. The case study and simulated results were used to evaluate the utility of each scoring approach.

3aSC16. Interactions among perceived breathiness, roughness, and overall severity of dysphonia. David A. Eddins, Supraja Anand (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu), and Rahul Shrivastav (Vice President for Instruction, Univ. of Georgia, Athens, GA)

Dysphonic voices typically co-vary across multiple quality dimensions. This study investigates the interaction between breathiness and roughness and their combined contributions to judgments of overall dysphonia severity. Four dysphonic talkers were replicated using a Klatt synthesizer with the Liljencrants-Fant (LF) model as the glottal excitation source. Two vowel continua were created for each speaker by systematically varying aspiration noise (AH) and open quotient (OQ) [to vary the magnitude of breathiness] and the waveform amplitude modulation depth [to alter the magnitude of roughness]. The stimulus matrix for each talker comprises 49 stimuli (7 breathy levels X7 rough levels) resulting in a total of 196 stimuli (4 talkers X49 stimuli).Ten naïve listeners provided judgments of breathiness, roughness, and overall dysphonia severity using a magnitude estimation task. Analyses determined the interaction between talker and listener judgments of breathiness, roughness, and overall severity. A set of iso-severity curves from the ME task were derived and illustrated the combination of breathiness and roughness magnitude to dysphonia severity. These data will be useful in establishing the validity of clinical scales for voice quality perception. [Work supported by NIH DC009029.]

3aSC17. Differential impact on speech production from chronic thalamic DBS in Essential Tremor and that from STN DBS in Parkinson's disease. Emily Wang (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Ste. 1017 AAC, 600 South Paulina, Chicago, IL 60612, emily_wang@rush.edu) and Leonard A. Verhagen Metman (Neurological Sci., Rush Univ. Medical Ctr., Chicago, IL)

Essential Tremor (ET) is the most common tremor syndrome seen in adults. The characteristic tremor in ET is postural and action tremors, with a frequency of 4–7 Hz; while in patients with Parkinson's disease (PD), the dominant tremor is resting tremor with a typical frequency of 5 Hz. Deep brain stimulation (DBS) in the ventral intermediate nuclei (Vim) has been shown effective in treating action or intention tremors in medically resistant ET, while DBS in the subthalamic nucleus (STN) has been shown effective in treating rigidity, rest tremor and Dopa-induced dyskinesias in PD. However, chronic bilateral DBS, whether in Vim or STN, may have negative impact on speech production. We report two cases where both patients received DBS greater than one year and developed speech impairment. When the stimulation in Vim was turned off for 30 min in ET, the patient's speech returned to the baseline while turning the STN stimulation off for 30 min in PD, minimal changes were observed and 12 h with stimulation-off were required for speech to return to the non-stimulated state.

3aSC18. Variations of F2 range measures in mild dysarthria and healthy talkers. Michaela McLaughlin and Lynda Feenaughty (School of Commun. Sci. and Disord., The Univ. of Memphis, 4055 N Park Loop, Memphis, TN 38152, mmclghl2@memphis.edu)

Diminished vocalic segments indexed by the extent, duration, and rate of change in the second formant (F2) are common in dysarthria and are known to reduce intelligibility in adults with multiple sclerosis (MS; Hartelius *et al*., 2010). When F2 range was measured over select sentences from a reading passage, results indicated that the F2 range may not be sensitive to

mild dysarthria in MS (Rosen *et al*., 2008). However, measures of the F2 range should be further explored for sentences imposing greater speech-motor and cognitive-linguistic demands. As such, phonetic contexts requiring even small vocal tract movement may contribute to intelligibility for speakers with mild dysarthria in MS. Thus, this study will investigate the impact of sentence length and complexity on within-speaker variations of F2 movement and group differences for 12 adults with mild dysarthria in MS and 12 healthy talkers. Various F2 range measures shown to be sensitive to speech motor capabilities will be obtained from 11 audio-recorded sentences varying in length and complexity. The results of this study will provide a better understanding of the speech motor control capabilities of speakers with mild dysarthria with relatively preserved intelligibility in MS. [Work supported by ASH Foundation and the University at Buffalo MDRF.]

3aSC19. Response times of repeated nonwords in children with residual speech sound disorder. Caroline E. Spencer, Sisan Cuervo, Natalie Baldinelli, Caroline Sheehan, and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., P.O. Box 670379, Cincinnati, OH 45267, spenceco@mail.uc.edu)

Nonword repetition (NWR) is sensitive to impairments in a range of disorders, including speech sound disorders (Larrivee and Catts, 1999). NWR requires listeners to perceive an auditory stimulus, store the perception in working memory, and repeat the stimulus aloud when prompted. Response time—from stimulus presentation to participant response—can be measured to assess processing speed. Deficits in each of these processes have been linked to speech sound errors in young children (Bird, *et al*., 1995; Lewis, *et al*., 2011; Preston, *et al*., 2015; Rvachew, *et al*., 2003), but research from older children with *residual* speech sound disorders (RSSD) is lacking. To evaluate processing speed in children with RSSD, we have adapted a nonsense Syllable Repetition Task (SRT; Shriberg *et al*., 2009). The SRT consists of multisyllabic nonsense words that use targeted phonemes (/b, d, m, n, ɑ/), to minimize the possibility of articulation difficulties masking the skills intended to be studied—phonological processing. Response times on 2-, 3-, and 4-syllable nonwords will be compared between children with RSSD and children with typically developing speech. Based on preliminary findings, we suspect that response times of children with RSSD are slower than those of children with typical speech. The results will be discussed.

3aSC20. Conversational turn-taking in youth at clinical high-risk for psychosis. Laura Sichlinger (King's College London, London, United Kingdom), Emily Cibelli, Matthew Goldrick, and Vijay Mittal (Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 608, matt-goldrick@northwestern.edu)

Previous studies of speech abnormalities in schizophrenia reflect a consensus that atypical turn-taking behavior during conversation is a prominent feature of speech patterns in psychosis. However, the literature focuses on the population with formal psychosis and lacks studies that involve individuals who are at clinical high-risk (CHR) for developing a psychotic disorder. This study assessed the turn-taking performance of CHR youth to evaluate if abnormalities in turn-taking can be observed in the CHR population compared with controls and to investigate if turn-taking performance is correlated with symptom severity. While analysis of speech data from structured clinical interviews shows no significant group effects, high scores in attenuated psychosis symptoms were associated with significantly longer between turn pauses in the CHR population. This study adds to a growing body of literature that suggests that patterns of speech and linguistic features may be useful in detecting risk, and tracking clinical and treatment course across schizophrenia and spectrum disorders.

3aSC21. Predicting children's word recognition accuracy with two distance metrics. Emma Brown (Dept. of Speech and Hearing Sci., Indiana Univ., 0 S. Jordan Ave., Bloomington, IN 47405, emebrown@indiana.edu), Izabela A. Jamsek, Laura Liang, Rachael F. Holt (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH), and Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Children generally have more difficulty in recognizing words produced by talkers with unfamiliar regional dialects and nonnative accents compared to their home dialect, but the specific deviations from native norms that cause these word recognition decrements have not been quantified. This study examines the relation between word recognition accuracy and two distance metrics: phonemic pronunciation distance and holistic perceptual distance. To calculate phonemic pronunciation distances, sentences from six speakers representing different regional dialects (British and Scottish English) and nonnative accents (Japanese-, German-, Mandarin-, and Hindi-accented English) were phonemically transcribed and compared to transcriptions from speakers of the ambient dialect (Midland American English). Preliminary results show a relation between the phonemic pronunciation distances and children's word recognition accuracy for the sentences. Follow-up work will measure holistic perceptual distances through a ladder task in which listeners will rank a larger set of native and nonnative talkers based on the perceived distance to the home dialect, a measure that will capture both segmental and suprasegmental differences. Comparing the two distance metrics will provide mechanistic insight into the impact of extent and type of talker deviation from native norms on children's word recognition accuracy. [Work supported by National Science Foundation #17570 and Indiana University Hutton Honors College.]

3aSC22. The (minimal) influence of perceived age on the perceived gender of children's speech. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

The acoustic characteristics of boys and girls' speech resemble adult norms for men and women, respectively, early in life. Moreover, the extent to which a child's speech resembles the adult norms for their biological sex is related to other measures of the extent to which their gender development meets cultural norms [e.g., Perry *et al*., *J. Acoust. Soc. Am.* **109** (01); Munson *et al*., *J. Acoust. Soc. Am.* **137** (15); and Beaulieu *et al*., *J. Acoust. Soc. Am.* **143** (18)]. The acoustic cues to gender in children's speech are distributed throughout the speech signal. Hence, the most common method to determine the attainment of gendered speech is by gathering perceptual ratings of gender typicality. In this study, we examine whether ratings of the gender typicality of children's speech are affected by ratings of another potentially salient attribute that speech acoustics convey: speaker age. We conducted an experiment in which perceived gender and perceived age ratings were collected for the single-word productions of 57 5–13 year old children examined by Beaulieu *et al*. (18) and brief spoken narratives by the same children. Preliminary results suggest that the effects of talker sex are robust when both actual age and perceived age are controlled statistically.

3aSC23. Perception of prosody: How toddlers harness intonation to guide early attention. Jill C. Thorson (Commun. Sci. and Disord., Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Dover, NH 03824, jill.thorson@unh.edu)

Young infants are born with language-specific preferences, particularly with respect to prosody (i.e., melody and rhythm). Previous work has shown that by 18-months, toddlers are guided by intonation and information status during an on-line reference resolution task (Thorson and Morgan, 14). This

study isolated the role of fundamental frequency (f0) during early attentional processing, showing that a bitonal f0 movement increases looking time to a target over a monotonal movement (and both show increased looking versus no pitch movements). The motivation for the current study is to examine the ability to perceive and utilize specific intonational patterns at earlier stages in speech and language development. The study asks whether typically developing 14-month-old toddlers are able to employ different intonational contours in order to attend to an object with unique information statuses (e.g., *new*, *given*). Methods include monitoring eye movements in response to varying pitch patterns and analyzing variables such as *total fixation time to a target* and *time of first fixation*. We hypothesize that at this early stage, toddlers will exploit the prosodic system to fixate on the discourse salient target. Critically, this work is a precursor to analyzing the early perception of prosody in young children with autism spectrum disorders.

3aSC24. Analysis of tongue trajectory variation for /r/ in older children and adults. Sarah Dugan (Psych., Univ. of Cincinnati, 67 Grady Court, Dayton, OH 45409, hamilsm@ucmail.uc.edu), Colin Annand (Psych., Univ. of Cincinnati, Cincinnati, OH), Sarah R. Li, Hannah M. Woeste, Jack A. Masterson (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Michael A. Riley (Psych., Univ. of Cincinnati, Cincinnati, OH), T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Kinematic studies of tongue articulator movement have found that older children and adolescents have smaller articulator displacements and greater movement variability compared to adults. However, many of these kinematic studies have used simple, early-developing phonemes as stimuli for comparison across talker groups. The American English rhotic /r/ is the latest to be acquired in children, possibly due to the complexity of the pharyngeal and oral constriction pattern required to produce the sound. In this study, we use automated processing of midsagittal ultrasound images to track trajectories of tongue blade, dorsum, and root during /r/ production in older children and adults. Our preliminary results suggest that the trajectories of tongue parts are different for adults and children, with children producing tongue part movements with greater relative amplitude and variability. We explore the possible developmental reasons for this outcome through the lens of dynamic principles of stability and change.

3aSC25. Secondary acoustic cues in adult perception of young children's stop productions. Elaine R. Hitchcock (Commun. Sci. and Disord., Montclair State Univ., 116 West End Ave., Pompton Plains, NJ 07444, hitchcocke@mail.montclair.edu) and Laura L. Koenig (Haskins Labs., New Haven, CT)

The perception of plosive voicing distinctions is attributed primarily to voice onset time (VOT). However, perceptual judgments of naturally produced stop tokens may be influenced by secondary cues, such as fundamental frequency (f0) and burst amplitude. The present study assessed the role of secondary cues in adult labeling judgments of bilabial and alveolar CV words produced by 2–3-year-old English-speaking children. Child productions were divided into three groups across the VOT continuum. Four exemplars per phoneme category (/p, b, t, d/) in each range (short, ambiguous, and long) were chosen from 6 children. Two stimulus sets were created. One set included pre-voiced tokens for /b d/ and exaggerated long lag tokens (100+ ms) for /p t/; the other set removed these extreme tokens. Listening data were obtained from adults per set. Listeners were highly accurate (>90%) at labeling the production as intended by the child when the production matched the VOT category expected for the target word. When VOT was ambiguous, judgments for voiced targets ranged from 75% to 92% and voiceless targets ranged from 54% to 62% accuracy. Measures of f0 and burst amplitude suggest that secondary cues in the speech signal contributed to adults' perception of children's stops.

3aSC26. Classification of accurate and error tongue movements for /r/ in children using trajectories from ultrasound. Sarah R. Li (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, CVC, 3960, Cincinnati, OH 45241, lisr@mail.uc.edu), Sarah Dugan (Psych., Univ. of Cincinnati, Dayton, OH), Colin Annand (Psych., Univ. of Cincinnati, Cincinnati, OH), Hannah M. Woeste, Jack A. Masterson (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Michael A. Riley (Psych., Univ. of Cincinnati, Cincinnati, OH)

American English /r/ is considered the most difficult sound to remediate in children with speech sound disorders. The sound is complex, requiring coordinated, quasi-independent tongue movements, making it challenging to remediate in speech therapy. However, there are few clinical tools that allow for a quantitative, real-time display of the dynamics of tongue part movements. We aim to fill this need by developing a real-time, simplified, interactive biofeedback system that transforms ultrasound image data to a visual feedback object and quantitative data stream. In this presentation, we compare blade, dorsum, and root displacement trajectories determined from midsagittal ultrasound imaging of children with residual speech sound disorders (RSSD) and children with typical speech, using principal component models and cluster analysis techniques. Results show a trend of smaller tongue part displacements for RSSD children compared to children with typical speech. The analysis also elucidates distinct strategies for production of accurate /r/. We compare our classification of accurate and error tongue movements to perceptual judgments from trained listeners. Preliminary results suggest strong correspondence between our trajectory-based classification and listener judgments of /r/ accuracy.

3aSC27. Photomicrography of the middle ear ossicles. Matthew Kist and Peter M. Scheifele (Audiol., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267, kistmj@mail.uc.edu)

I removed temporal bone sections from human cadavers with the help of my colleagues. From there, I utilized a Dremel tool to access the otic capsule and remove the malleus, incus, and stapes from each section. Not all attempts were successful due to the fragile nature of these bones. I extracted a total of 5 stapes, an incus, and 2 mallei in total. These bones were photographed using photomicrography: a 35 mm film camera attached to a microscope. The purpose of this research is to provide high quality images of the middle ear ossicles for educational purposes and learning tools. The ossicles will also be present in addition to the poster for individuals to hold and accurately grasp the size of these bones.

Session 3aSP

Signal Processing in Acoustics, Underwater Acoustics, Acoustical Oceanography, and Noise: Bayesian Inference in Acoustic Signal Processing

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

Zoi-Heleni Michalopoulou, Cochair
Mathematical Sciences, New Jersey Institute of Technology, 323 ML King Blvd, Newark, NJ 07102

Paul J. Gendron, Cochair
ECE Department, University of Massachusetts Dartmouth, 285 Old Westport Road, North Dartmouth, MA 02747

Chair's Introduction—8:30

Invited Papers

8:35

3aSP1. Physics-based distributions for randomly scattered signals and their Bayesian conjugate priors. D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr.-CRREL, 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), Vladimir E. Ostashev, Daniel J. Breton, and Carl R. Hart (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Sound waves propagating through the atmosphere and ocean are randomly scattered by turbulence, internal waves, surface roughness, and other processes. In some limiting cases, probability density functions (pdfs) for the scattered signal variations can be derived, e.g., the log-normal pdf for weak scattering (in the Rytov approximation) and the exponential pdf for strong scattering. A variety of more general, usually empirically based, distributions are available which reduce to these limiting cases, such as the Rice, gamma, and generalized gamma. For situations involving multiple receivers, multivariate log-normal, Wishart, and matrix gamma pdfs may be employed. Parametric uncertainties and spatial/temporal variability in the scattering process can be addressed with a compound pdf formulation, which involves an additional distribution for the uncertain or variable parameters. From a Bayesian perspective, the scattering pdf corresponds to the likelihood function, the pdf for the uncertain parameters to the prior/posterior, and the compound pdf to the marginal likelihood. Many common scattering pdfs possess Bayesian conjugate priors, which lend themselves to simple updating equations and analytical solutions for the posteriors and marginal likelihoods. This presentation summarizes important pdfs for randomly scattered signals and their conjugate priors when available.

8:55

3aSP2. Bayesian inference when the physics is not quite right. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., P.O. Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Bayesian inference requires a physical and structural model for the portion of the environment being assessed. The model is characterized by a finite number of parameters, and these are assumed to each be a random variable. The forward problem can be solved given a knowldege of these parameters and the governing equations. Bayesian inference begins (prior probability) with some initial broad assumptions about the probability distributions. An intricate (Bayesian) process involving data and solutions to the forward problem leads to a refinement of these probability distributions. The range of possibilities becomes narrower, and one identifies most probable values of the parameters. The present paper raises the question as to what results when the parameterization is capriciously in conflict with what is known about the actual environment. An example from underwater acoustics is the choice of parameters to describe the frequency dependence of attenuation on a sediment layer. What is sometimes done is to assume the attenuation is directly proportional to frequency (just one parameter) or that the attenuation obeys a power law with a constant exponent (two parameters). The intrinsic shear modulus of the sediment is ignored in the numerical solutions of the forward problem, so the physics is incomplete. Some idealized examples are used to explore the consequences of such simplifying assumptions. Improved parameterizations are suggested that will yield more realistic frequency dependences of attenuation.

9:15

3aSP3. Trans-dimensional geoacoustic inversion and uncertainty quantification for SBCEX17 data. Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Jan Dettmer (Univ. of Calgary, Calgary, AB, Canada), Charles W. Holland (Appl. Phys. Lab., The Penn State Univ., State College, PA), Josee Belcourt (School of Earth & Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Julien Bonnel (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Dag Tollefsen (Defence Systems Div., Norwegian Defence Res. Establishment, Horten, Norway), and David P. Knobles (Knobles Sci. and Anal., Austin, TX)

This paper presents an efficient and general approach to Bayesian inversion and uncertainty quantification for seabed geoacoustic profile estimation. The model-selection problem of estimating an appropriate seabed parameterization is addressed with trans-dimensional (trans-D) inversion via reversible-jump Markov-chain Monte Carlo, which samples probabilistically over the number of layers. An efficient proposal density for parameter perturbations is based on using a linearized approximation to the posterior probability density, applied in principal-component (PC) space where the (rotated) parameters are uncorrelated. The acceptance rate of perturbations and birth/death steps is improved by parallel tempering, based on a series of interacting Markov chains with successively tempered (relaxed) likelihoods. The PC proposals are adapted individually to the tempering of each Markov chain. The data-error model is based on the assumption of multivariate Gaussian errors with correlations represented by an autoregressive process. The parameters of zeroth- and first-order autoregressive error processes are sampled trans-dimensionally to avoid over- or under-parameterizing the error model. The approach is illustrated for three data sets from the 2017 Seabed Characterization Experiment (SBCEX17), including broadband seabed reflection coefficients; dispersion of water-borne acoustic modes, resolved by warping analysis; and ship noise recorded at a bottom-mounted horizontal array of hydrophones.

9:35

3aSP4. Employing geometric pattern discovery techniques for sonar target recognition. Ananya Sen Gupta (Elec. and Comput. Eng., Ctr. for the Eng. Arts and Sci., Univ. of Iowa, 4016 Seamans Iowa City, IA 52242, ananya-sengupta@uiowa.edu) and Ivars P. Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

Automated sonar target recognition suffers from linear superposition of acoustic color features, which need to be precisely disambiguated to correctly identify a target of the specific composition and geometry. Most target acoustic color images exhibit feature overlap between environmental reverberation as well as reflections due to target geometry, and elastic waves that are representative of the material composition of the target. From a signal processing perspective, this is essentially mitigating and disentangling interference from two highly competing yet hard-to-predict sources: reflections from the environment and reflections due to the target geometry, which represent different physical phenomena with distinct geometric patterns. We will present recent results in feature selection and interference mitigation employing Gabor wavelets to separate target acoustic color features due to elastic wave orbits from interfering features due to the target geometry and environmental effects. We will provide results from simulations as well as experimental field data and discuss the trade-offs in feature representations employing acoustic color and those that also include the phase information. Finally, we will discuss how this type of feature extraction can be used to discover and create robust priors for Bayesian inference networks for learning salient target features.

9:55

3aSP5. Hybrid Cramer-Rao bound for near-field source localization using a spatially spread acoustic vector sensor. Zakayo Ndiku Morris (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., Hung Hom KLN, Hong Kong, zakayo.morris@connect.polyu.hk), Kainam Thomas Wong (none, San Gabriel, CA), and Yuting Han (Data Sci. Program, Worcester Polytechnic Inst., Worcester, MA)

This presentation characterizes the source-localization precision obtainable via received signal strength indication (RSSI) based on data from a tri-axial velocity sensor and a spatially separated pressure sensor. That scheme was proposed originally by Y. I. Wu and K. T. Wong in January 2012 in the *IEEE Transactions on Aerospace and Electronic Systems*. That source-localization scheme depends on the acoustic propagation path-loss exponent, which is typically not precisely known a priori but could be modeled stochastically. That exponent may also differ in value for the propagation path to the tri-axial velocity sensor and for the propagation path to the pressure sensor. This presentation accounts for these two practical considerations in characterizing the scheme's source-localization precision, through the metric of the hybrid Cramer-Rao bound (HCRB), the correctness of which is here validated by Monte Carlo simulations of the corresponding maximum a posteriori (MAP) estimator.

10:15–10:30 Break

10:30

3aSP6. Bayesian inference for water column and sediment inversion in ocean acoustics. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

Bayesian methods provide inference on unknown parameters given prior beliefs and a statistical model for the observations in a noisy environment. It was shown in "Sequential filtering and linearization for inversion in the Seabed Characterization Experiment," *J. Acoust. Soc. Am.* **144**, 1913 (2018) how geometry and sediment parameter estimation in an oceanic experiment can be achieved with a two-stage Bayesian approach. Here, we demonstrate the potential of the method for inversion with a different data set presenting challenges in identifying features of the acoustic field necessary for the second stage of the estimation process. The water column sound speed is complex in this experiment, necessitating a complex inverse model. The transmitted signals are lfm pulses and probability densities of multipath arrival times are calculated at the receivers. For the forward and inverse calculations, we compute the Jacobian matrix for sound speed estimation using EOF coefficients. The inversion, employing the direct and surface and bottom paths, leads to inference on the geometry and the water column sound speed profile. Using the computed probability densities and introducing one more arrival, sediment property densities are calculated. Results are in excellent agreement with expected values for the site. [Work supported by ONR.]

10:50

3aSP7. Localization of a mobile acoustic scatterer from sub-Rayleigh resolved acoustic arrivals in a refractive environment. Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, North Dartmouth, MA) and Abner C. Barros (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, abarros1@umassd.edu)

Computational Bayesian inference on the state of an underwater mobile object from a continuous active acoustic transmission is presented. The challenge of sub-Rayleigh resolvable wave vectors in refractive environments with uncertainty in ambient acoustic noise power is addressed. The location and speed of the mobile scatterer are inferred under the challenging constraint of a small receive vertical aperture. The need for jointly inferring the vertical angles and Doppler offsets of the arrivals is addressed with a Gibbs sampling approach. The posterior density of the plane wave components is mapped to the object's range, depth, and speed through ray interpolation. A case scenario from an acoustic duct environment in the western Indian ocean is presented.

11:10

3aSP8. Bayesian framework for direction of arrival analysis using spherical harmonics. Stephen Weikel (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180, weikes@rpi.edu), Christopher Landschoot (Kirkegaard Assoc., Chicago, IL), and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

A common task in acoustical applications is the determination of directions of arrival (DoAs) of sound at a receiver. This work aims to address this problem in situations involving potentially multiple simultaneous sound sources by means of a two-level framework of Bayesian inference. This process involves first estimating the number of sound sources present, followed by estimating their directional information, based on sound data collected with a spherical microphone array. Analytical models are formulated using spherical harmonic beamforming techniques, which are incorporated into the Bayesian analysis as part of the prior information. The experimental data are also incorporated to update the information available prior to analysis. All necessary prior information is assigned based on the principle of maximum entropy. Through this technique, the number of sources is first estimated, and then, the DoA information of those sources is extracted from the most concise model that adequately fits the experimental data. This paper presents the Bayesian formulation and analysis results to demonstrate the potential usefulness of model-based Bayesian inference for determining DoAs in complex noise environments with potentially multiple concurrent sources.

Contributed Paper

11:30

3aSP9. Sparse Bayesian learning beamforming combined with short-time Fourier transform for fault detection of wind turbine blades. Xiang Pan, Zefeng Cheng, Zheng Zheng, and Yehui Zhang (School of Information and Electron. Eng., Zhejiang Univ., Zhe Rd. 38, Hangzhou, Zhejiang 310027, China, panxiang@zju.edu.cn)

Wind energy is one of the important renewable energy resources. The wind turbines need to be checked every now and then to enhance security. The rotor blade of the wind turbine can be damaged due to long-term running in harsh environments and complex vibrations resulting in the crack of blade. For detection of the blade faults, the sparse Bayesian learning (SBL) beamforming (BF) is implemented to the acoustic data received by a microphone array on the ground for signal enhancement. The direction of arrival of the abnormal sound can be estimated with high resolution meanwhile interferences, such as noise emitted by cooling fans, can be suppressed by the low sidelobes provided by the SBL-BF. After the Short-Time Fourier Transform (STFT) is carried out over the enhanced signals, it is seen from the time-frequency spectrum that the abnormal sound appears with an approximate 6-s cycle. By detecting the cyclical characteristics, one can decide whether there is or not the blade fault. A real-time processing system for detection of the blade faults underwent a number of tests in a coastal plain, a hilly area, and the Xinjiang Plateau. The test results have demonstrated the effectiveness of the blade fault detection framework.

Session 3aUW

Underwater Acoustics and Acoustical Oceanography: Ocean Acoustics in High Latitudes and General Propagation

Mohsen Badiey, Chair
University of Delaware, University of Delaware, Newark, DE 19716

Contributed Papers

9:00

3aUW1. Azimuthal dependence of the acoustic field in a year long Canada Basin Acoustic Propagation Experiment. Mohsen Badiey (Univ. of Delaware, University of Delaware, Newark, DE 19716, badiey@udel.edu), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Sean Pecknold (Defence Res. and Development Canada, Dartmouth, NS, Canada), Megan S. Ballard, Jason D. Sagers (Appl. Res. Lab.Univ. of Texas, Austin, TX), Altan Turgut (Naval Res. Lab., Washington, DC), John A. Colosi (Oceanogr., Naval Post Graduate School, Monterey, CA), Peter F. Worcester, and Mathew Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

As in the mid-latitudes, the variability of the oceanography in the arctic ocean causes azimuthal dependent acoustic fields induced by the environmental parameters, such as the variable sound speed profile and bathymetry. Preliminary analysis of a year-long experimental data from "deep-to-shallow" and the "shallow-water" Canada Basin Acoustic Propagation Experiment (SW CANAPE) shows strong azimuthal variability of broadband signals recorded on 11 spatially distributed acoustic receiver arrays on the Chukchi shelf from September 2016 to October 2017. Particular attention is paid on geotimes pertaining to the variable sound speed profiles around the dynamic shelf-break region. Although acoustic modeling of the propagation requires a physics based ocean model for a complete modeling exercise, preliminary determination of the azimuthal behavior of broadband propagation is conducted using multiple two-dimensional Parabolic Equation (Nx2D) with the collected oceanographic data. Calculations for long-range, as well as short-range, source-receiver pairs in the SW CANAPE data show that azimuthal variabilities occur frequently and are related to the spatial and temporal characteristics of signal dispersion for different range and geotime scales. [Work supported by ONR.]

9:15

3aUW2. Temporal and spatial dependence of a yearlong record of sound propagation from the Canada Basin to the Chukchi Shelf. Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Mohsen Badiey (Univ. of Delaware, Newark, DE), John A. Colosi (Naval Postgrad. School, Monterey, CA), Altan Turgut (U.S. Naval Res. Lab., Washington, DC), Sean Pecknold (Defence Res. and Development Canada, Dartmouth, NS, Canada), Ying-Tsong Lin, Andrey Y. Proshutinsky, Richard A. Krishfield (Woods Hole Oceanographic Inst., Woods Hole, MA), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

During the Canada Basin Acoustic Propagation Experiment (CANAPE), low-frequency signals from five tomographic sources located in the Canada Basin were recorded by an array of hydrophones located on the Chukchi Shelf. The propagation distances ranged from 240 km to 520 km, and the propagation conditions changed from persistently ducted in the basin to seasonally upward refracting on the continental shelf. An analysis of the received level from the tomography sources revealed a spatial dependence in the onset of the seasonal increase in transmission loss, which was correlated with the locations of the sources in the basin. This observation led to the hypothesis that the water advected from Barrow Canyon westward over the continental slope by the Chukchi slope current contributes to the temporal and spatial dependence observed in the acoustic record. The water column properties and ice draft were measured by oceanographic sensors on the basin tomography moorings and by six arrays of oceanographic moorings on the continental shelf to characterize the temporal and spatial variability of the environment. This talk examines the range-dependent measurements and explains the observed variability in the received signals through propagation modeling. [Work sponsored by ONR.]

9:30

3aUW3. Seabed properties on the Chukchi Shelf observed during the 2016–-2017 canada basin acoustic propagation experiment. Jason D. Sagers, Megan S. Ballard (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu), and Sean Pecknold (Defence Res. and Development Canada, Dartmouth, NS, Canada)

Previous work examined seabed geoacoustical properties near the 150 m isobath on the Chukchi shelf as observed during the 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE) [Sagers and Ballard, *JASA* **144**(3), 1666]. That work reported a water/sediment compressional sound speed ratio near 0.98 and a compressional sound speed gradient in the upper sediment around 6 s^{-1}. This work extends the prior analysis by investigating additional locations throughout the larger shallow-water experimental site. Data from subbottom profile surveys, measurements from an *in situ* acoustic coring system and inferences made using ship-radiated noise received on Autonomous Multichannel Acoustic Recorders (AMARs) deployed by the Defence Research and Development Canada (DRDC) are examined to understand seabed layering and sediment properties throughout the experimental region. Particular interest is given to understanding whether geoacoustical properties on the Chukchi shelf exhibit range-dependence. [Work sponsored by ONR.]

9:45

3aUW4. Modeling mid-frequency reverberation in the Arctic Ocean. Dajun Tang (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

With renewed interest in the Arctic in response to its changing environment, various acoustic field measurements are underway or being planned. Anticipating the utility of reverberation as a tool to probe the environment, as well as to detect targets, here a time-domain reverberation model suitable for the frequency band of 0.5—5 kHz is introduced. This model assumes that the sea ice as the scattering mechanism and sea ice scattering parameters are taken from a set of laboratory measurements. It is found that the double-duct sound speed profile in the Arctic enables sound to propagate to long distances, and the main loss mechanism is scattering by sea ice, including ice keels. Reverberation measured at different azimuths is hypothesized

as an effective method to estimate angular-dependent keel density and varying sound channels. [Work supported by ONR Ocean Acoustics.]

10:00

3aUW5. Forty-year review of developments in underwater acoustic modeling. Paul C. Etter (Northrop Grumman Corp., P.O. Box 1693, Baltimore, MD 21203, paul.etter@ngc.com)

This is the sixth paper in a series of reviews presented at eight-year intervals starting in 1979 [*J. Acoust. Soc. Am.* **65**, S42 (1979); **82**, S102 (1987); **97**, 3312 (1995); **114**, 2430 (2003); **129**, 2631 (2011)]. All surveys covered basic acoustic models and sonar performance models. Basic acoustic models included 143 propagation, 29 noise, and 32 reverberation models. Propagation models were categorized according to ray theory, multipath expansion, normal mode, fast field, and parabolic approximation formulations; further distinctions were made between range-independent and range-dependent solutions. Noise models included ambient noise and beam-noise statistics models. Reverberation models included cell and point-scattering formulations. The 44 sonar performance models included active sonar models, model operating systems, and tactical decision aids. Active sonar models integrated basic acoustic models, signal-processing models, and supporting databases into cohesive operating systems organized to solve the sonar equations. Model operating systems provided a framework for the direct linkage of data-management software with computer-implemented codes of acoustic models. Tactical decision aids represented a form of engagement-level simulation that blended environmental and sonar performance information with tactical rules. The overall inventory increased by 5 models per year; historical references are maintained in the fifth edition of *Underwater Acoustic Modeling and Simulation*.

10:15–10:30 Break

10:30

3aUW6. Shallow water acoustic propagation prediction. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell St., B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

A set of propagation loss curves, extracted from measured reverberation in an environment with water depths of 200–250 ft is used to investigate the crossover region between deep and shallow water using two normal mode implementations designed to predict propagation in deep and shallow water, respectively. A third range-recursive calculation which is applicable in situations for which cycle mixing with range results in randomization of phase interference between modes is also used for comparison. The impact of forward boundary scattering on reproducing the measured levels at multiple receiver depths is demonstrated.

10:45

3aUW7. Modeled position and intensity statistics in deep water caustics. Katherine F. Woolfe (Leidos, 672 Brookline St. SW, Atlanta, Georgia 30310, katherine.woolfe@gmail.com), Chuck Spofford (Leidos, Fairfax, VA), Kritika Vayur (Leidos, State College, PA), and Peter Mikhalvesky (Leidos, Arlington, VA)

Caustics are a form of natural focusing that occurs in deep water acoustic propagation, but acoustic intensity and position perturbations at caustics have not been investigated in detail. The positions, shapes, and intensities of caustics and cusps are controlled primarily by the large-scale sound speed profile, with perturbations primarily caused by deterministic features (fronts, eddies) and stochastic features (internal waves). We selected two representative deep water environments for our analysis: a north Pacific site and temperate profile. For each region, 100 independent and 100 time-coherent realizations of internal waves were generated. The model uses the statistics provided by the Garrett-Munk power spectrum to generate displacements, which are converted into sound speed realizations. In the north Pacific profile, internal waves were concentrated primarily in the upper 200 m of water. In the temperate profile, internal waves were concentrated in the upper 700 m of water. We used a Gaussian beam model to analyze position and intensity statistics of caustics for both types (independent and time-coherent) of internal wave fields at each site as a function of frequency, internal wave strength, and source depth. Simulations indicate that caustics are more stable than other portions of the acoustic field and that this stability is a function of internal wave strength and caustic location. This work is approved for public release, Distribution Unlimited.

11:00

3aUW8. Analytic solution to a waveguide featuring caustics and shadow zones. Brian M. Worthmann (Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550, bworthma@umich.edu)

Exact solutions to waveguides are useful tools for validating numerical waveguide models. In this paper, a modal solution to an unbounded waveguide with a range-independent but depth-dependent profile is described. The sound speed profile utilized is a symmetric, piecewise n^2-linear profile matched above and below to a homogenous half-layer with density and sound speeds that prevent any reflections at the interface. In this environment, caustics and shadow zones are formed. A nearly exact solution to this environment could help improve other numerical methods' accuracy near caustics and shadow zones. In this mathematical analysis, both proper (trapped) and improper (leaky) modes are included. Arbitrarily high frequencies are permitted through the use of numerically well-conditioned approximations to Airy functions, particularly for the dispersion relations and mode-shape evaluations. The proper (real) and improper (complex) modal eigenvalues are found approximately, with root-finding algorithms used for eigenvalues near cut-off to improve the approximation. Mathematical details are presented along with a brief discussion of how well a few common numerical solvers agree with the analytic solution derived.

11:15

3aUW9. Preliminary soundscape observations from a Colombian humpback whale breeding ground before port construction. Kerri Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Christina E. Perazio (Univ. at Buffalo, Biddeford, Maine), Valeria Gonzalez (Pacifico Adventures, Buenos Aires, Argentina), Andrea Luna-Acosta (Pontificia Universidad Javeriana, Bogota, Colombia), and Natalia Botero (Fundación Macuáticos Colombia, Medellin, Colombia)

The Gulf of Tribugá in the Colombian Pacific is still relatively undisturbed. No road access between villages nor from major urban areas exists. Small boats and walking beaches at low tide serve as the main transportation conduits. In this breeding ground for humpback whale Stock G, small-scale artisanal and shrimp fisheries and whale-watching activities support the livelihood of local communities but may already interfere with biological communication systems. Once a proposed international port is built anthropogenic pressure is expected to increase. Baseline information is key to understand the Gulf's current acoustic state and to document the best approximation of its practically pristine state. An ecological Acoustic Recorder (EAR, Oceanwide Science Institute) was deployed between the largest town (about 2000 residents) and the southern boundary of the Utria National Park marine reserve from October to December, 2018. Also, opportunistic over-the-side recordings from July to September, 2018, provided point surveys at a greater spatial resolution during peak humpback whale breeding season. Results include the first acoustic catalogue of the area and a preliminary understanding of important bandwidths for future monitoring and current contributions of small vessels to the soundscape.

11:30

3aUW10. Characterization of sound channel axis and depth in the global ocean. Mukunda Acharya and Likun Zhang (National Ctr. for Physical Acoust., Univ. of Mississippi, 112-, 114 Chucky Mullins Dr., Oxford, MS 38655 , mkachary@go.olemiss.edu)

Sound speed in the ocean has a minimum at a certain depth that acts as the axis of sound channels for underwater sound propagation. In this work, we characterize the three dimensional features of the axial sound speed and depth in the global ocean. The characterization follows from calculation of more than 16 000 sound speed profiles by using the conductivity, temperature, and pressure data in the World Ocean Circulation Experiment. The axial speed and depth as a function of latitude are determined for the Atlantic, Pacific, and Indian oceans, and a polynomial fitting for the dependence is provided. The variability of axial sound channel and depth between

different oceans and latitude is identified and related to the feature of the thermoclines of each ocean. The characterization and formulation provide knowledge of the sound channel in the global ocean that could be particularly useful for modeling long-range and three-dimensional underwater sound propagation for applications in underwater acoustic tomography and monitoring seismic activities.

WEDNESDAY AFTERNOON, 15 MAY 2019 FRENCH, 1:00 P.M. TO 2:35 P.M.

Session 3pAA

Architectural Acoustics: Acoustical Materials and Testing

Benjamin Bridgewater, Cochair
Architecture, University of Kansas, 1536 Ogden Street, Denver, CO 80218

Shane J. Kanter, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—1:00

Contributed Papers

1:05

3pAA1. Broadband design of multilayer micro-slit panel absorbers for improved transparency using Bayesian inference. Michael Hoeft, Cameron J. Fackler, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, hoeftm@rpi.edu)

Multilayer Micro-Slit panels (MSP) are assessed for their potential as broadband absorbers that simultaneously maintain visual transparency. The late Dah-You Maa introduced micro-Slit panel absorbers as a continuation of his previously developed Micro-Perforated panel (MPP) absorbers. Like MPPs, Micro-Slit panels allow high absorption coefficients to be achieved without using fibrous materials but are limited to a narrow frequency bandwidth. Broadband absorption can be achieved by combining panels into a multilayer absorber. The complexity of determining the optimum set of parameters required to fulfill a design scheme necessitates implementation of the Bayesian framework. This probabilistic method automatically determines the most concise number of layers required and gives parameters for each layer of the resulting composite. Various slit patterns and algorithms introduce a family of solutions, which satisfy the design scheme, for optimization of visual transparency. The Micro-Slit samples can be fabricated and tested to validate acoustic performance and assess visual properties.

1:20

3pAA2. Studies on sound absorption performance of porous materials at low temperature. Xiwei Wang (HuaXinSiFang (Beijing) Construction Technol. Co., Ltd., Beijing, China), Xiang Yan (Acoust. Lab of School of Architecture, Tsinghua Univ., Beijing, China), Hui Li (Beijing Deshang Jingjie Technol. Development Co., Ltd., Main Bldg., Rm. 104, Beijing 100084, China, lihuisylvia@aliyun.com), and Xiaoyan Xue (HuaXinSiFang (Beijing) Construction Technol. Co., Ltd., Beijing, China)

The acoustical parameters of porous materials are usually tested at normal temperature. In recent years, spaces need acoustical treatments at low temperature spring up with the dynamic tendency of projects such as snow theme parks and high speed trains traveling in the frozen region. The sound absorption coefficient of porous materials at low temperature is very rare for now. So in real projects, the sound absorption coefficient of porous materials at normal temperature is the only reference which is certainly inappropriate. The sound absorption coefficient of several common porous materials is test at −50°C, −25°C, and 25°C. The details of the measurement process and data analysis are specified in this paper.

1:35

3pAA3. The spectrum of concrete: Material research and experimentation expanding previous sound lab results for various acoustical environments. Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

This paper presentation continues progress of an active multiphase acoustical research project emphasizing implementation of prototypical concrete systems in multiuse spaces to transform unhealthy learning environments. Building upon research results shared during the December 2017 and November 2018 ASA paper presentations, various admixtures/additives are being explored as viable ingredients to develop concrete mixtures capable of decreasing excessive reverberation and flutter echo while increasing clarity and speech intelligibility. Faculty and students, working in both physical and digital realms with industry partners, are analyzing and validating consequences of manipulating materials for resultant acoustical criteria. The work presented here showcases research and development of how concrete can be processed, shaped, surfaced, and implemented into full-scale built form to provide scholarly merit applicable in/with similar spaces and/or functions.

1:50

3pAA4. Impact sound isolation measurement analysis for 6 typical type of floors in China. Xiaoyan Xue (HuaXinSiFang (Beijing) Construction Technol. Co., Ltd., Beijing, China), Hui Li (Beijing Deshang Jingjie Technol. Development Co., Ltd., Main Bldg., Rm. 104, Beijing 100084, China, lihuisylvia@aliyun.com), and Xiang Yan (Acoust. Lab of School of Architecture, Tsinghua Univ., Beijing, China)

Single layer homogeneous elastic material floating floor, porous vibration isolation material floating floor, vibration isolation cube floating floor, spring floating floor, floor with hard finish topping, and floor with soft finish

topping are 6 types of impact sound isolation floor commonly used in China. All the 6 types of floor are test in laboratory by both light impact method with standard tapping machine and heavy impact method with rubber tire and rubber ball. Data analysis is focused on one hand the difference between test methods for the same floor and the other hand the sound isolation performance of different floors by the same test methods.

2:05

3pAA5. A high-resolution goniometer to measure sound diffraction patterns. Anthony Savino, Jonathan Kawasaki, and Ning Xiang (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, savina@rpi.edu)

Surface scattering and diffuse reflections from acoustic diffusers have recently become a significant topic of research for room acoustics. An acoustical goniometer can be implemented to create a circular microphone array to characterize the polar response for various types of diffusers. Particular areas of interest include validating the Physical Theory of Diffraction (PTD) through the use of finite-sized diffusers. In order to properly describe this phenomenon, a microphone array must achieve a fine enough angular resolution in addition to fulfilling the far-field requirements. This talk will discuss the implementation and results of a portable goniometer with a radius of 5 m and an angular resolution of 1.25 deg. This design is intended to be easily deployed in empty, indoor spaces of sufficiently large dimensions for a full 360-deg diffraction response measurement.

2:20

3pAA6. Adaptive reuse of unconventional spaces for performance: Music in a museum and opera in a warehouse. Samuel H. Underwood (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE), Kieren H. Smith (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Brigham Young University, Provo, UT 84602, kierenhs@gmail.com), and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE)

Taking a space designed for one purpose and creating an acoustically sound space for musical performance presents unique challenges. A local opera company has undertaken two projects that involve such modifications for this performance season and have sought guidance on the unique acoustic complications involved. The first project was a percussive musical playground in a large art museum, adjacent to sound-sensitive exhibits. The second involved the adaptation of an existing warehouse into a space suitable for the performance of a ballet-opera. Researchers worked with set designers early in the process and throughout set construction. In order to better understand the spaces, impulse responses were taken and analyzed. Impulse response measurement results, acoustic models with suggested modifications, and auralizations at key listening positions are presented to address unique challenges associated with these unconventional spaces. An emphasis has been placed on reaching target values for reverberation time within limited time and resource constraints. Additionally, sound isolation for the two spaces has been addressed with special regard to adjacencies and noises in industrial downtown locations.

WEDNESDAY AFTERNOON, 15 MAY 2019 CLEMENTS, 1:30 P.M. TO 2:45 P.M.

Session 3pAB

Animal Bioacoustics: Topics in Animal Bioacoustics

Christopher Biggs, Chair
Marine Science Institute, University of Texas, 750 Channelview Dr., Port Aransas, TX 78373

Contributed Papers

1:30

3pAB1. Bald eagles (*Haliaeetus leucocephalus*) monitor their immediate acoustic environment vigilantly. JoAnn McGee, Peggy B. Nelson, Jeff Marr, Julia Ponder, Christopher Milliren, Christopher Feist, Andrew Byrne, and Edward J. Walsh (Univ. of Minnesota, S39 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, mcgeej@umn.edu)

As part of an initial effort to determine if acoustic signals can be used to discourage eagles from entering wind turbine facility airspaces and thereby reduce morbidity/mortality collision rates, behavioral responses of bald eagles (*Haliaeetus leucocephalus*) to a battery of both natural and synthetic acoustic stimuli of varying spectral complexities were studied. Each signal was directed randomly to one of two loudspeakers in a sequence of ~10 trials, and the stimulus order was randomized for each bird. A variant of an observer-based psychoacoustic protocol was implemented, and judges were instructed to report the absence or presence of a response, response strength, and other distinctive response attributes. In pilot studies, subjects responded to ~74% of trials across all stimuli. Responsivity was greater to spectrally complex stimuli (~80% vs 59%), and greater responsivity was observed to natural stimuli than to synthetic stimuli (~82% vs 69%). Responsive subjects oriented correctly in the direction of the signal source in ~74% of trials. A significant difference in overall responsivity was not observed across stimulus sets, although habituation was observed across repeated trials when responses to all stimulus types were combined. The relevance of findings in relation to the design of deterrence/alerting protocols will be discussed. [Work supported by DOE Grant No. DE-EE0007881.]

1:45

3pAB2. The effects of age and sex on rates of hearing loss for pure tones varying in duration. Anastasiya Kobrina and Micheal L. Dent (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

Mice are frequently used to study and model presbycusis due to similarities in the human and mouse cochleae and in genetic makeup. Most of the previous research on presbycusis used electrophysiological measures of hearing in mice, leading to an underrepresentation of behavioral experiments in the literature on mouse aging. The goal of the current research was

to fill this gap by behaviorally measuring audiograms and temporal summation functions in aging mice. Adult mice were trained and tested using an accelerated longitudinal design. Mice were trained on a detection task using operant conditioning procedures with positive reinforcement. Audiograms were constructed using thresholds for 8, 16, 24, 42, and 64 kHz pure tones. Temporal summation functions were constructed for 16 and 64 kHz pure tones ranging from 20 to 800 ms in duration. The results revealed that mice retain pure tone hearing late into their lifespan, with high-frequency hearing loss preceding low-frequency hearing loss. Mice also benefit from increases in the duration of pure tones; however. this benefit decreases with age. Generally, male mice lose hearing at a faster rate than females. These results highlight the importance of measuring hearing in awake, trained, behaving subjects when comparing presbycusis across species.

2:00

3pAB3. Arabian horse vocalization: A brief look at romance. David Browning (Browning Biotech, 139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com), Peter Herstein (Browning Biotech, Westerly, RI), and Peter M. Scheifele (Univ. of Cincinnati, Cincinnati, OH)

Arabian horses are an expressive breed. It has been determined that domestic horse whinnies have two unrelated fundamental frequencies, termed F(0) and G(0), with their harmonics such as F(2), etc. F(0) can be connected to arousal, while G(0) to mood. Fifteen Arabian whinnies during quiet barn conditions were analyzed to determine typical calm F(0) and G(0) values. A limited number of whinnies were than obtained from a corralled stallion during his recognition, anticipation, and culmination of a mare being brought to nuzzling distance on the other side of the fence. The spectra of his prenuzzling whinnies are as might be expected but after the culmination he gives a loud whinny with a complex structure that is not totally understood.

2:15

3pAB4. Propagation loss of spawning calls produced by spotted seatrout *Cynoscion nebulosus* and the effective detection area of passive acoustic sampling. Christopher Biggs (Marine Sci. Inst., Univ. of Texas, 750 Channelview Dr., Port Aransas, TX 78373, cbiggs@utexas.edu), Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, Austin, TX), and Brad Erisman (Marine Sci. Inst., Univ. of Texas, Port Aransas, TX)

Acoustic signaling in fish has been observed in conjunction with various behaviors but most commonly is associated with spawning. Sound production has been used to identify spawning sites for multiple species, but sound propagation characteristics are often not accounted for, severely limiting the spatial resolution at which spawning sites can be identified. We examined the propagation loss of seatrout calls in the very shallow environment of an estuary using recorded calls and an array of stationary hydrophones. We estimated the minimum expected source level of a seatrout call from the maximum recorded sound pressure level (SPL) of individual fish calls made *in situ* at spawning sites. Based on preliminary data from 86 samples, the minimum expected source level of an individual call is 142.1 ± 0.9 (CI_{95}) dB_{rms} *re*: 1 μPa at 1m. Measurements of propagation loss were fitted with a logarithmic model, and preliminary results showed that seatrout calls were detectable above background noise levels between 64 m and 512 m from the source. These results can be used to relate sound levels to the abundance of fish present at a spawning aggregation and enhance the precision of locating spawning sites using passive acoustics.

2:30

3pAB5. Sex-differences in timing of the black-capped chickadee fee-bee song. Anastasiya Kobrina (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu), Allison H. Hahn (Psych., St. Norbert College, Edmonton, AB, Canada), Eduardo Mercado (Psych., SUNY Univ. at Buffalo, Buffalo, NY), and Christopher B. Sturdy (Psych., Univ. of Alberta, Edmonton, AB, Canada)

Male black-capped chickadees produce *fee-bee* songs in spring for mate attraction and territorial defense. Less is known about the female song use in this species although a version of song, soft-song, appears to be used in mate-mate communication. Recent analyses of songs produced by chickadees revealed that female *fee-bee* songs are distinct from male songs in the spectral domain. Chickadees also precisely control the timing of their *fee-bee* songs during territory defense. No previous work has explored whether there are sex differences in the temporal patterning of *fee-bee* song production. Inter-song intervals were extracted from recordings of *fee-bee* songs produced in non-social contexts by 8 male and 7 female birds. Male chickadees produced *fee-bee* songs regularly, with the majority of songs spaced at intervals of 2.5–5.0 s. Song timing by females was more variable with production intervals ranging from 1.5 to 8.0 s. The relative stereotypy of song timing by males is consistent with earlier work suggesting that males may modulate song timing to communicate with other birds (e.g., by timing song production to either reduce or increase the likelihood that their songs overlap with those of other singers), and with differential use of songs by males and females.

Session 3pBAa

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Competition (Poster Session)

Kevin J. Haworth, Chair
University of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45209

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD $500 for first prize, USD $300 for second prize, and USD $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee.

Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions associated with the abstract numbers. All entries will be on display, and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

2aBA2. Live color encoded speckle imaging platform for real-time complex flow visualization *in vivo*
Student author: Billy Y. S. Yiu

2aBA3. Design of carotid bifurcation phantoms for integrative imaging investigations of arterial wall and flow dynamics
Student author: Adrian J. Y. Chee

2aBA4. Atherosclerosis characterization using lipid-specific photoacoustic imaging and 4D ultrasound strain mapping in mice
Student author: Gurneet S. Sangha

2aBA5. Spatial analysis of cardiac strain using high-frequency four-dimensional ultrasound in mice
Student author: Frederick William Damen

2aBA6. Quantification of murine cardiac hypertrophy using 4D ultrasound
Student author: Alycia G. Berman

2aBA10. Ascertaining the relationship between acoustic droplet vaporization, inertial cavitation, and hemolysis
Student author: Newsha Jahanpanah

2aBA11. Frequency dependence of the vaporization threshold of sonosensitive perfluorocarbon droplets varying their liquid core and size
Student author: Mitra Aliabouzar

2aBA12. Acoustic droplet vaporization with microfluidic droplets results in dissolved oxygen scavenging
Student author: Rachel P. Benton

2pBA4. Detection of nucleic acid-loaded microbubbles in mouse hearts during ultrasound-mediated delivery
Student author: Meghan R. Campbell

2pBA5. Pentagalloyl glucose effects on murine abdominal aortic aneurysms
Student author: Jennifer L. Anderson

3aBA4. Spectroscopic photoacoustic imaging for cardiovascular interventions
Student author: Sophinese Iskander-Rizk

3pBAb5. Evaluation of a potential medical diagnosis application of sonoluminescence
Student author: Alicia Casacchia

3pBAb6. Deep-learning framework and acoustic reflector for improved limited-view and sparse photoacoustic tomography
Student author: Irvane Ngnie Kamga

3pBAb7. Pixel-wise deep learning for improving image reconstruction in photoacoustic tomography
Student author: Steven Guan

4aBAa7. Super-resolution ultrasound imaging for in vivo microvasculature assessment in acute kidney injury mouse model
Student author: Qiyang Chen

4aBAa8. Efficient sub-diffraction passive cavitation imaging
Student author: Scott J. Schoen

4aBAa10. Echo-mode aberration tomography: Sound speed imaging with a single linear array
Student author: Anthony Podkowa

4aBAa13. Inferring elastic moduli of drops in acoustic fields
Student author: Jesse Batson

4aBAb1. A parametric evaluation of shear wave speeds estimated with the time-of-flight approach in viscoelastic media
Student author: Luke M. Wiseman

4aBAb2. Measured power law attenuation of shear waves in swine liver
Student author: Steven A. Grosz

4aBAb4. Approximate analytical time-domain Green's functions for space-fractional wave equations
Student author: Madison Carriere

4aBAb5. Validity of independent scattering approximation (ISA) to measure ultrasonic attenuation in porous structures with mono-disperse random pore distribution
Student author: Yasamin Karbalaeisadegh

4aBAb7. Investigation into tendon histotripsy
Student author: Molly Smallcomb

4aBAb9. Design of a histotripsy array for the treatment of intracerebral hemorrhage
Student author: Tyler Gerhardson

4aBAb11. *Ex vivo* thermal ablation monitoring using three-dimensional ultrasound echo decorrelation imaging
Student author: Elmira Ghahrahmani Z.

4pBAa1. Photoacoustic tomography in a clinical linear accelerator for quantitative radiation dosimetry
Student author: David A. Johnstone

4pBAa2. Comparisons of inverse and forward problem approaches to elastography
Student author: Siavash Ghavami

4pBAb5. Microstructural anisotropy evaluation in trabecular bone structure using the mode-converted (longitudinal to transverse, L-T) ultrasonic scattering
Student author: Omid Yousefian

4pBAb9. Standing acoustic waves in microfluidic channels for enhanced intracellular delivery of molecular compounds
Student author: Connor S. Centner

4pBAb12. Development and characterization of acoustically responsive exosomes for simultaneous imaging and drug delivery applications
Student author: Jenna Osborn

4pBAb14. Optimization of molecular delivery to red blood cells using an ultrasound-integrated microfluidic system
Student author: Emily Margaret Murphy

5aBA1. Electroacoustic tomography system using ultra-short electric filed excitation source induced acoustic signals
Student author: Ali Zarafshani

5aBA3. Tissue Doppler imaging to detect muscle fatigue
Student author: Joseph Majdi

5aBA7. Loudness growth as a function of presentation method: Comparison of normal hearing children with children using cochlear implants
Student author: Shubha Tak

5aBA9. Renal volume reconstruction using free-hand ultrasound scans
Student author: Alex Benjamin

5aBA10. Etiology of the color Doppler twinkling artifact on kidney stones
Student author: Scott A. Zinck

5aBA11. The effect of crystal chemical composition on the color Doppler ultrasound twinkling artifact
Student author: Eric Rokni

Session 3pBAb

Biomedical Acoustics, Signal Processing in Acoustics and Physical Acoustics: Interaction of Light and Ultrasound II

E. Carr Everbach, Cochair
Engineering, Swarthmore College, 500 College Avenue, Swarthmore, PA 19081

Ronald A. Roy, Cochair
Dept. of Engineering Science, University of Oxford, 30 Edgeway Road OX3 0HD, United Kingdom

Invited Papers

1:20

3pBAb1. Antivascular photo-mediated ultrasound therapy and its application in the eye. Xinmai Yang (Mech. Eng., Univ. of Kansas, 1530 West 15th St., Leaned Hall 3138, Lawrence, KS 66045, xmyang@ku.edu), Yannis M. Paulus (Dept. of Ophthalmology and Dept. of Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), and Xueding Wang (Dept. of Biomedical Eng. and Dept. of Radiology, Univ. of Michigan, Ann Arbor, MI)

Antivascular therapy can improve the prognosis of a variety of pathological conditions, including cancer and many eye diseases. By synergistically applying laser pulses and ultrasound bursts, we developed a photo-mediated ultrasound therapy (PUT) technique as a localized antivascular method. PUT takes advantage of the high native optical contrast among biological tissues and has the unique capability to self-target microvessels without damaging surrounding tissue. The technique utilizes an integrated therapeutic ultrasound and laser system. The laser system emits 5-ns, 10-Hz pulses, which is synchronized with 10-ms, 10-Hz ultrasound bursts. Experiments were carried out on chicken yolk sac membrane and rabbit eyes. With radiant exposures around 5 mJ/cm^{2}at 532 nm and ultrasound pressures around 0.4 MPa at 1 MHz or 0.5 MHz, microvessels were able to be removed. Furthermore, *ex vivo* tests with human blood demonstrated that cavitation was induced when laser and ultrasound were utilized synergistically. On the rabbit eye model, the blood flow in microvessels could be greatly reduced after PUT, and the occlusion of microvessels could last up to 4 weeks. In conclusion, PUT holds significant promises as a non-invasive method to precisely remove microvessels in neurovascular eye diseases by more selectively treating vasculature with minimized side-effects.

1:40

3pBAb2. Optical mean irradiance change for inference of focused-ultrasound-induced strain and thermal expansion in tissue-like materials. R. Glynn Holt (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, rgholt@bu.edu), Ali Vakili, Joseph L Hollmann, and Charles A. DiMarzio (Comput. and Elec. Eng., Northeastern Univ., Boston, MA)

Inferring material properties in scattering media, especially tissue, is a commonly applied method for detecting flaws or abnormalities. Hybrid acoustic/optic methods are often employed to overcome the limitations inherent in one or the other. Such hybrid methods have achieved great success in imaging various features (mechanical or optical properties) at a variety of spatial scales down to the subcellular. Here, we seek to develop an intermediate resolution method of detecting changes in material properties that is robust, inexpensive, and potentially capable of real-time analysis. We exploit the ability of the time-averaged absorption of focused ultrasound to induce changes in the optical index of scattering materials. By combining simulations with experiments, we demonstrate that the optical mean irradiance change measurement is capable of revealing time-dependent index changes. We are able to separate the contributions of both thermal expansion and radiation force deformation in the correlation signal. Potential applications of this technique will be discussed.

2:00

3pBAb3. Combining light and sound with nanoparticles to identify and treat head and neck cancers. Teklu Egnuni (Leeds Inst. of Medical Res., St. James' Univ. Hospital, Leeds, United Kingdom), Li Chunqi (School of Electron. and Elec. Eng., Univ. of Leeds, Leeds, United Kingdom), Nicola Ingram, Louise Coletta (Leeds Inst. of Medical Res., St. James' Univ. Hospital, Leeds, United Kingdom), Steven Freear, and James R. McLaughlan (School of Electron. and Elec. Eng., Univ. of Leeds, University of Leeds, Leeds LS2 9JT, United Kingdom, J.R.McLaughlan@leeds.ac.uk)

High intensity focused ultrasound (HIFU) is a non-invasive and non-ionising approach used primarily for the thermal ablation of cancerous tumours. Even though it has been in clinical use for over 30 years, it has yet to achieve widespread use. Two key limitations for this approach are long treatment times and a difficulty in getting real-time feedback on treatment efficacy. One technique that could help with these limitations is a combination of HIFU, pulse laser illumination, and cancer targeted nanoparticles. When nanoparticles are simultaneously exposed to these modalities, vapour bubbles form, providing a controllable way to nucleate cavitation in the target location. Acoustic emissions from inertial cavitation can be monitored via passive cavitation detection and/or mapping. This approach provides direct localisation of cancerous regions and has greater sensitivity compared with current photoacoustic imaging. Once the

cancerous regions have been localised, they can be ablated by HIFU, which is known to be enhanced in the presence of cavitation, by enhancing thermal damage in a localised region. Furthermore, the acoustic emissions generated during these ablations could give an indication of treatment progress. This study will present data on both *in vitro* and *in vivo* validation of this approach in models of head and neck cancer.

Contributed Papers

2:20

3pBAb4. HIFU tissue lesion quantification by optical coherence tomography. Jason L. Raymond (Dept. of Eng. Sci., Univ. of Oxford, 17 Parks Rd., Oxford OX1 3PJ, United Kingdom, jason.raymond@eng.ox.ac.uk), E. Carr Everbach (Eng., Swarthmore College, Swarthmore, PA), Ronald Roy (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Manuel Marques, Michael Hughes, and Adrian Podoleanu (School of Physical Sci., Univ. of Kent, Canterbury, Kent, United Kingdom)

Heating of tissue by high-intensity focused ultrasound (HIFU) can result in sufficient temperature elevation to cause irreversible changes in the tissue structure. The contiguous volume occupied by these changes, a lesion, and the extent of the tissue changes may be quantified histologically or estimated through techniques such as ultrasonic elastography. We have shown that changes in tissue optical scattering could be used as a proxy to improve sensing and imaging of HIFU lesion formation as an alternative to thermometry. Optical coherence tomography (OCT) is a light-based method appropriate for optically accessible tissues, which we have used to quantify lesion volume, shape, and quality based upon the irreversible changes in optical scattering that occurs with protein denaturation. We have adapted OCT to take into account changes in optical polarization of the tissue, providing sensitivity to changes in the collagen orientation of skin with heating. This technique has potential in detecting antecedents of skin burn during HIFU exposures, thereby increasing safety and reducing treatment times.

2:35

3pBAb5. Evaluation of a potential medical diagnosis application of sonoluminescence. Alicia Casacchia (Walker Dept. Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St., Austin, TX 78712, acasacchia@utexas.edu), Parker George (Plan II Honors Program, Univ. of Texas at Austin, Austin, TX), Preston S. Wilson, and Mark F. Hamilton (Walker Dept. Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Sonoluminescence (SL) is a phenomenon in which light is produced via violent collapse of a gas-filled cavity under excitation by a varying pressure field. Although the precise mechanism of this light production is not yet agreed upon in the literature, various applications of this effect have been established and are continually being developed. One such example of an application of this phenomenon that has not been extensively studied is the production of SL in biological fluids as a method for medical diagnostics. Measurements performed by Chernov *et al*. [in Proceedings of 14th International Symposium on Nonlinear Acoustics (1996), pp. 219–223] revealed varied intensities of SL emissions in blood plasma samples from groups of patients diagnosed with different diseases. We present an experimental apparatus for the production of single bubble sonoluminescence (SBSL) and subsequent measurement of radial oscillations using optical scattering techniques. This system will allow for characterization of the effects of both the biological fluid content and viscoelasticity on the spectra of SBSL light emissions. Additionally, these experimental results can be used to inform future computational models of this behavior. [This is a Plan II SAWIAGOS project.]

2:50

3pBAb6. Deep-learning framework and acoustic reflector for improved limited-view and sparse photoacoustic tomography. Irvane Ngnie Kamga, Steven Guan, and Parag V. Chitnis (BioEng., George Mason Univ., 12300 Oak CreenREEK Ln., Apt. 1009, Fairfax, VA 22033, ingnieka@gmu.edu)

Photoacoustic imaging is a hybrid imaging modality that relies upon optical absorption of pulsed light and subsequent thermoelastic generation of ultrasound. The detection of the induced acoustic waves outside the tissue enables image reconstruction. A major challenge encountered in photoacoustic tomography (PAT) lies in the inability to acquire complete projection data from the region of interest due to both the limited view and sparsity of available ultrasonic sensors. The resulting images are characterized by severe artifacts and poor quality. In this work, we examined the utility of incorporating an acoustic reflector to address the limited view problem and to train a convolutional neural network (CNN) to improve PAT image reconstruction from sparsely sampled data. Photoacoustic wave propagation was simulated in MATLAB using the k-Wave toolbox. We compared the performance of a sparse linear transducer array (with and without reflector) to that of a circular transducer array. The structural similarity index (SSI) was used as a metric for evaluating image quality. The combination of a curved reflector and artifact-removal using a CNN improved the quality of PAT images from the linear configuration. The resulting mean SSI value (0.859) was comparable to that achieved using the circular transducer array (0.926).

3:05

3pBAb7. Pixel-wise deep learning for improving image reconstruction in photoacoustic tomography. Steven Guan, Amir Khan, Siddartha Sikdar, and Parag V. Chitnis (BioEng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, Sguan2@gmu.edu)

Photoacoustic tomography involves absorption of pulsed light and subsequent generation of ultrasound, which when detected using an array of sensors can produce clinically useful images. Practical considerations limit the number of sensors and their "view" of the region of interest (ROI), which can result in significant reconstruction artifacts. Iterative-reconstruction methods can improve image quality but are computationally expensive. Another approach to improve reconstructed images is to use convolution neural networks (CNN) as a post-processing step for removing artifacts. However, missing or heavily obscured features typically cannot be recovered using this approach. We present a new pixel-wise deep learning (PDL) approach that employs pixel-wise interpolation to window ROI-specific raw photoacoustic data and then directly performs the image reconstruction within the CNN framework. The utility of this approach was demonstrated on simulated photoacoustic data from a 64-element semi-circular sensor array. The training and testing datasets comprised of 500 images from a synthetic vasculature phantom and 50 images of an anatomically realistic vasculature obtained from micro-CT images, respectively. The structural similarity index of the PDL-reconstructed images (0.91 ± 0.03) indicated superior image quality compared to those obtained using the iterative reconstruction (0.82 ± 0.09) and CNN-based artifact removal (0.79 ± 0.07).

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Christina J. Naify, Chair
Acoustics, Jet Propulsion Lab, 4800 Oak Grove Dr, Pasadena, CA 91109

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Hot topics in computational acoustics. D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr.-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil)

As reflected by the formation of a new Computational Acoustics Technical Specialty Group in the ASA, computational methods attract acoustics researchers and practitioners across the spectrum of the current ASA Technical Committees. This presentation samples several prominent hot topics in computational acoustics: (1) high-resolution, three-dimensional solutions for sound fields in the ocean, atmosphere, and indoor spaces, (2) time-domain treatments of dissipative processes and reactive boundary conditions, (3) machine learning and other data-driven surrogate models trained with more computationally intensive physics-based models, and (4) characterization of uncertainty in computations and its quantification through methods such as adaptive sampling and polynomial chaos.

1:35

3pID2. Acoustic metamaterials and additive manufacturing. Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

Acoustic metamaterials (AMM) have captured the attention of a wide range of researchers for their potential to create materials with properties or functionality that exceed naturally occurring materials or conventional composites. Progress in acoustic metamaterials is largely the product of a combination of technological advances that have been made in the past three decades. Specifically, AMM research is a prototypical example of novel concepts in physics converging with advances in technology, primarily additive manufacturing and widespread access to robust computational tools. Additive manufacturing is a key component to this area of research since it allows for rapid build and test cycles as well as the production of elaborate structures for acoustic wave manipulation that follow from rigorous mathematical predictions. This talk will highlight physical concepts central to AMM research and the role that advanced manufacturing processes play in present and future research. It will then discuss the significant technical challenges that must be overcome if AMM are to be brought to their full potential, including the need to fabricate large amounts of subwavelength dynamic microstructure spanning orders of magnitude in length scale.

2:05

3pID3. Emerging imaging methods in biomedical ultrasound. T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Possibilities for diagnostic and therapeutic ultrasound imaging methods are continually extended by advances in transducer and beamformer technology, algorithms for signal processing and image reconstruction, research-oriented imaging platforms, and computing power. This talk describes and illustrates recent advances in biomedical ultrasound imaging methods from multiple groups. In different respects, each provides information beyond the tissue reflectivity, blood flow, and strain mapped by current clinical ultrasound scanners. Methods discussed include quantitative imaging of tissue parameters by solving inverse scattering and inverse elasticity problems, super-resolution imaging beyond the diffraction limit, and passive imaging of acoustic cavitation. Physical acoustics principles underlying the methods, as well as applications in diagnosis and therapy guidance, are discussed.

Session 3pMU

Musical Acoustics, Psychological and Physiological Acoustics and Signal Processing in Acoustics: Polyphonic Pitch Perception and Analysis II

Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

M. Torben Pastore, Cochair
Architectural Acoustics, Rensselaer Polytechnic Institute, 4 Irving Place, Troy, NY 12180

Invited Papers

1:00

3pMU1. Monaural edge pitch and models of pitch perception. William M. Hartmann (Phys. and Astronomy, Michigan State Univ., Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824, hartmann@pa.msu.edu), Peter Cariani (Hearing Res. Ctr., Biomedical Eng., Boston Univ., Newton, MA), and H. Steven Colburn (Hearing Res. Ctr., Biomedical Eng., Boston Univ., Boston, MA)

Noise with a sharp spectral edge produces a definite, perceived pitch. The pitch of lowpass noise lies slightly below the edge and the pitch of highpass noise slightly above it. The pitch shifts away from the edge, expressed in semitones, increase with the decreasing edge frequency. A neural timing model based on positive peaks of the autocorrelation function accounts for the broad features of the shifts. So, does a place model based on the peak of an excitation pattern as enhanced by lateral inhibition. As the edge frequency decreases below 150 Hz, the pitch persists for a lowpass noise but disappears for highpass noise. This result is consistent with the timing model but not with the place model. For high edge frequencies, the pitch is stronger for highpass noise than for lowpass noise—consistent with both timing and place models. As the edge frequency approaches 5000 Hz, the pitch disappears for most listeners, but for some listeners, a pitch persists for edge frequencies beyond 8000 Hz. The latter result argues against a timing model. It appears that both timing processes (low edge frequency) and place processes (high edge frequency) are required to explain the edge pitch. [Work supported by the AFOSR and the NIDCD.]

1:20

3pMU2. Decoding MEG responses to musical pitch reveals the dynamic emergence of tonal structure in human cortex. Narayan Sankaran (Neurological Surgery, Univ. of California, San Francisco, 675 Nelson rising Ln., San Francisco, CA 94158, narayan.sankaran@ucsf.edu), Thomas A. Carlson (Psych., The Univ. of Sydney, Sydney, NSW, Australia), and William Forde Thompson (Psych., Macquarie Univ., Sydney, NSW, Australia)

Tonal music is characterized by a hierarchical structuring of pitch, whereby certain tones appear stable and others unstable within their musical context. Despite its prevalence, the cortical mechanisms supporting such a percept remain poorly understood. We examined the neural processing dynamics underlying pitch-structure in Western Tonal Music. Listeners were presented with tones embedded within a musical context whilst their magnetoencephalographic (MEG) activity was recorded. Using multivariate pattern analysis, decoders attempted to classify the identity of tones from their corresponding MEG activity at each peristimulus time-sample, providing a dynamic measure of their cortical dissimilarity. Time-evolving neural distinctions were then compared with the predictions of several acoustic and perceptual models. Following the onset, a temporal evolution was witnessed in the representational structure in cortex. While MEG dissimilarities between tones initially corresponded to their fundamental frequency separation, distinctions beyond 200 ms reflected their status within the hierarchy of perceived stability. Transposing dissimilarities corresponding to this latter period into different keys, neural relations between keys correlated with the well-known *circle of fifths*. Convergent with fundamental principles of music-theory and perception, results detail the dynamics with which the complex perceptual structure of Western tonal music emerges in human cortex within the timescale of an individual tone.

Contributed Paper

1:40

3pMU3. Automated Mandarin tone classification using deep neural networks trained on a large speech dataset. Jeramey Tyler, Huang Zou (Cognit. Sci., Rensselaer Polytechnic Inst., 1709 Tibbits Ave., Troy, NY 12180, tylerj2@rpi.edu), Helen Zhou (Commun. and Media, Rensselaer Polytechnic Inst., Troy, NY), Hui Su (IBM, Troy, New York), and Jonas Braasch (Rensselaer Polytechnic Inst., Troy, NY)

A tonal language is one in which the speaker's intonation modifies the meaning of a word. In this work, we perform a rigorous analysis of intonation changes, or pitch contours, produced by native Mandarin speakers to predict the tone-contour type. Pitch contours are estimated using a number of different methods, also measuring each contour's Mel-Frequency Cepstral Coefficients (MFCCs). The dataset used was autonomously generated from the Aishell open-source Mandarin speech corpus. Each sample was aligned with its transcript using Montreal Forced Alignment and segmented into individual words. The resulting corpus covers 11 topic domains, spoken by 400 individuals. Separate development, training, and testing datasets are created to ensure the integrity of our results. Pitch contours and their MFCCs are exposed to a number of machine learning techniques including clustered, regression, and traditional Deep Neural Network (DNN) approaches. MFCCs are additionally processed using convolutional neural networks. The models are used to predict the corresponding tone for a contour. Our work seeks to determine which intonation representations perform optimally for machine learning tasks. The tool is used to provide audio and visual feedback to learners of tonal languages. [Work supported by RPI Seed Grant and CISL].

Invited Paper

1:55

3pMU4. Multiple f0 pitch estimation for musical applications using dynamic Bayesian networks and learned priors. David A. Dahlbom and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, dahlbd@rpi.edu)

The identification of multiple simultaneous pitches is a challenging signal processing task and cannot at present be performed as well as trained human subjects. Moreover, it appears that successful human performance depends on skill acquisition and knowledge of musical conventions. Even human capabilities are likely fairly poor in the absence of training and musical context. We present a framework, using Dynamic Bayesian Networks, that permits the principled incorporation of models of music theory, musical instruments, and human pitch perception. A particular advantage of this approach is the ability to develop each of these models independently, relying either on expert knowledge or machine learning as necessary. Models of appropriate complexity can then be selected for a specific application. In the present work, we focus on the use of learned models of musical context, specifically Deep Markov Models, and use these to improve inferences about simultaneous pitches. The main drawback of this framework is the intractability of the inference computations and the computational expense of approximation methods. We explore particle filtering as an approach to addressing these problems with the ultimate aim of making a system useable in a musical performance system. [Work supported by NSF BCS-1539276 and IBM AIRC grant.]

Session 3pNS

Noise, Architectural Acoustics, and Structural Acoustics and Vibration: Noise at Sporting Events and Sports Venues

Daniel A. Russell, Cochair
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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—1:20

Invited Papers

1:25

3pNS1. Impulsive exposures from ceremonial and signal cannons. Gregory Flamme (SASRAC, 2264 Heather Way, Forest Grove, OR 97116, gflamme@sasrac.com), William J. Murphy, Chucri A. Kardous, and David C. Byrne (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH)

Cannons, small firearms, and starter pistols are sometimes used with blank charges during sporting events, ceremonies, and historical re-enactments. The sound levels produced by such devices are not widely known, and it is possible that the personnel discharging them could underestimate the potential risk to hearing. Depending upon the proximity to participants and spectators, the sound levels produced by such devices can be potentially hazardous. The exposures have not been widely examined because they fall outside of regulations and standards that cover typical occupational or military exposures. This presentation describes the acoustic characteristics and exposure limits for two large-caliber ceremonial cannons and a signal cannon. The cannons produced impulses between 150 and 174 dB peak sound pressure level (SPL) and 8-h equivalent A-weighted levels ranging between 60 and 108 dBA, respectively. In addition, measurements from small firearms and starter pistols are presented, which produced sound levels between 145 and 165 dB SPL. Such sound levels exceed the various occupational recommended and permissible exposure limits. This paper provides recommendations for noise and administrative controls for all personnel within 15 m of the ceremonial cannons and 10 m of the signal cannon. Double hearing protection should be required during all activities.

1:45

3pNS2. Referee whistles Part I—Permissible exposures indoors. Trevor W. Jerome (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, MS 3220B, State College, PA 16804-0030, twjerome@gmail.com), Gregory Flamme (SASRAC, Forest Grove, OR), and William J. Murphy (Div. of Appl. Res. and Technol., National Inst. for Occupational Safety and Health, Cincinnati, OH)

Sound from referee whistles at sporting events is usually relatively short in duration (<250 ms) but generated at relatively high levels (>110 dB SPL). Damage risk criteria (DRC) that categorize potentially harmful sounds are usually meant for either continuous or impulsive noise. These types of whistle sounds are better categorized as impulsive. Measurements were taken of a trained referee using a sample of commercially available whistles in a controlled environment. One microphone was placed at the ear of the referee, and another was placed 1 m in front of the referee. Whistle signals were analyzed with a maximum duration of 450 ms, which would be typical of a sporting event. DRC for these impulsive sounds have been investigated using DRC of A-weighted 8-h energy (LeqA8), and the warned and unwarned conditions of the AHAAH model. Depending on the risk model used, the numbers of permissible exposures was as low as zero and extended above.

2:05

3pNS3. Referee whistles Part II—Outdoor sound power assessment. William J. Murphy (Hearing Loss Prevention Team, Ctr. 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), Stephen M. Tasko (Speech Pathol. & Audiol., Western Michigan Univ., Kalamazoo, MI), Donald Finan, Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), Michael Stewart (Dept. of Commun. Disord., Central Michigan Univ., Mount Pleasant, MI), James E. Lankford (School of Allied Health and Communicative Disord., Northern Illinois Univ., Dekalb, IL), Adam R. Campbell (Hearing Loss Prevention Team, Ctr. for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH), and Gregory Flamme (SASRAC, Forest Grove, OR)

Referee whistles have been suggested as a significant contributor to noise-induced hearing loss. Thirteen models of sport whistles were tested for sound power with a 3-m hemispherical array of 19 microphones. The whistler produced nine tweets of low, medium, and high effort with two samples of each whistle model. Sound power levels ranged between 74 and 115 dB re 1 pW. The low, medium, and high effort tweets had average power levels of 85 $\pm$ 6 dB, 100 $\pm$ 6 dB, and 110 $\pm$ 4 dB, respectively. Preliminary damage-risk analysis of the whistle impulses yield varied estimates for the allowable number of tweets before auditory damage might be expected. For the Auditory Hazard Assessment Algorithm, between 4 and 66 tweets may exceed the daily exposure threshold. Based upon the amount of eight-hour equivalent A-weighted energy, approximately 120 to 500 tweets would exceed the daily 85 dBA exposure limit. The directivity of the sound power measurements will also be examined, and risk of hearing loss will be discussed.

2:25

3pNS4. Improved automated classification of basketball crowd noise. Mylan R. Cook, Brooks A. Butler, Katrina Pedersen, Spencer Wadsworth, Eric Todd, Kent L. Gee, Mark K. Transtrum, and Sean Warnick (Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com)

This paper describes using both supervised and unsupervised machine learning (ML) methods to improve automatic classification of crowd responses to events at collegiate basketball games. This work builds on recent investigations by the research team where the two ML approaches were treated separately. In one case, crowd response events (cheers, applause, etc.) were manually labeled, and then, a subset of the labeled events were used as a training set for supervised-ML event classification. In the other, (unsupervised) k-means clustering was used to divide a game's one-twelfth octave spectrogram into six distinct clusters. A comparison of the two approaches shows that the manually labeled crowd responses are grouped into only one or two of the six unsupervised clusters. This paper describes how the supervised ML labels guide improvements to the k-means clustering analysis, such as determining which additional audio features are required as inputs and how both approaches can be used in tandem to improve automated classification of crowd noise at basketball games.

Session 3pPA

Physical Acoustics, Engineering Acoustics, and Biomedical Acoustics: Acoustofluidics II

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Charles Thompson, Cochair
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Chair's Introduction—1:00

Contributed Papers

1:05

3pPA1. Simultaneous trapping and pulling by acoustical Bessel beams as stable tractor beams. Xudong Fan and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, University, Oxford, MS 38677, xfan1@go.olemiss.edu)

A stable tractor beam for long range pulling of particles by acoustic Bessel beams requires a simultaneous trapping in the lateral direction. Trapping force acting on a sphere centered on the axis of Bessel beams is examined to guide the selection of material and beam parameters. It is found that a heavy and rigid sphere in the Rayleigh regime can be trapped at central pressure maximum of zero-order Bessel beam, which is contrary to the repelling behavior at the pressure anti-nodes of one- or two-dimensional standing waves. The trapping results from the three-dimensional features of the velocity fields and the momentum projection. The projection leads the trapping of an elastic sphere or a droplet at the axis of ordinary and vortex Bessel beams in the Rayleigh regime to be reversed when reducing the conical angle of the beam at the situation when there is a strong contrast of mass density between the particle and the surrounding fluid. Ranges of conical angle and material parameters for simultaneous trapping and pulling a spherical object are identified.

1:20

3pPA2. Acoustic streaming in a channel a moderate streaming Reynolds number. Charles Thompson (ECE, UMASS, 1 University Ave., Lowell, MA 01854, charles_thompson@uml.edu), Jairo Vanegas (ME, UMASS Lowell, Lowell, MA), Russell Perkins, Flore Norceide, Ivette Alvarez, and Kavitha Chandra (ECE, UMASS, Lowell, MA)

In this work, the generation of acoustic streaming in a rigid walled channel is examined. At low values of the streaming Reynolds number, the time-averaged fluid motion in the channel follows that given by Rayleigh. However, departure from the aforementioned result ensues as the magnitude of the streaming Reynolds number increases. Higher order nonlinear corrections to the Rayleigh streaming solution is given and are expressed in terms of a regular perturbation sequence in nondimensional particle amplitude. It is shown that the reduction in the amplitude of the axially directed streaming velocity is a function of the streaming Reynolds number.

1:35

3pPA3. Numerical modeling of submicron particle acoustic concentration in gaseous flow. Jizhou Liu (School of Energy and Power Eng., Beihang Univ., No. 37 Xueyuan Rd., Beijing 100191, China, jizhou.liu@buaa.edu.cn), Xiaodong Li (School of Energy and Power Eng., Beihang Univ., Beijing, China), and Fang Q. Hu (Mathematics and Statistics, Old Dominion Univ., Norfolk, VA)

The acoustic concentration of submicron particles in a micro-channel is investigated numerically via a gas-particle multiphase coupling scheme. For modeling the gas flow under transverse standing wave, 2 dimensional linearized Euler equations with the parabolic mean flow are employed with the high order finite difference method. Through the analogous behavior of rarefied gas and air-suspended particles, a modified Unified Gas-Kinetic Scheme (UGKS) is adopted to estimate the particle dynamics. In detail, Stokes' drag force and acoustic radiation force applied on particles are accounted for by introducing a velocity-dependent kinetic acceleration term in the UGKS. To validate this modeling, numerical simulations are tested with varying standing wave amplitudes. The effects of acoustic radiation force to drag force ratio and mean flow velocity are also analyzed. The computed concentration patterns and efficiencies are compared with experimental results from the literature. The agreement shows that the adopted Euler-UGKS coupling scheme could be an effective tool for micro-sized particle acoustic concentration applications.

1:50

3pPA4. An acoustics separation technique based on the development of an interface in the acoustic field. Krishna N. Kumar, Adrian Barber, Jack Saloio, Tyler Campbell, Kedar C. Chitale, Benjamin P. Ross-Johnsrud (Res. & Development, FloDesign Sonics, Inc., 380 MainSt., Wilbraham, MA 01095, k.kumar@fdsonics.com), and Bart Lipkens (Res. & Development, FloDesign Sonics, Inc., Springfield, MA)

Chimeric antigen receptor (CAR) T-cell therapy is a promising and evolving immunotherapy approach for cancer treatment. In allogeneic CAR-T therapies, TCR + cells must be removed from the final cell product because of immunogenicity problems. It is accomplished through a negative affinity cell selection process where TCR + cells are affinity bound to a

bead. The harvested TCR-cells are the product cells. A multidimensional acoustic standing wave field separates cell-bead complexes from free cells in an acoustic fluidized bed. The feed solution motion is normal to the primary acoustic field. Irrespective of the particle acoustic contrast, an interface between a dense suspension on the bottom and clear fluid on top develops in the field. We examine the physics behind the development of the interface and its subsequent motion. This motion influences the purity, scalability, and recovery of the TCR- cells. We present the effects of different acoustofluidic parameters, e.g., bead concentration, bead acoustic contrast factor, frequency, and flow rate on interface formation and its movement. Theoretical calculations and experimental results are discussed. The acoustic fluidized bed has been shown to give final purities of 99 + % of TCR- cells from a starting purity of 60%–70%, with 70 + % recoveries of TCR- cells.

2:05–2:20 Break

2:20

3pPA5. Droplet extraction and manipulation at a fluid interface using fraxicon modified ultrasound. Robert L. Lirette (Phys., Univ. of Mississipi, 2400 Anderson Rd., Apt. 4, Oxford, MS 38655, rlirette@go.olemiss.edu), Joel Mobley (Phys., Univ. of Mississipi, University, MS), and Likun Zhang (Phys., Univ. of Mississipi, Oxford, MS)

Ultrasound focused at a fluid-fluid boundary creates an acoustic radiation pressure on the boundary that is dependent on the incident energy density and the relative density and sound speed of each fluid. For different fluid combinations, this radiation pressure can either be positive or negative. For this study, ultrasound propagating from water to carbon tetrachloride was used to create a negative radiation pressure at the interface. This fluid combination is impedance matched eliminating reflections and heating effects at the boundary. A fraxicon phase plate lens is a low profile analog of an axicon and generates an approximate Bessel beam in the far field. The near field exhibits a complex diffraction pattern including shadow zones capable of acoustic trapping. Starting with a planar interface, we demonstrate the extraction, capture, and manipulation of a carbon tetrachloride droplet. The negative radiation pressure draws the carbon tetrachloride surface up into the water, eventually breaking a droplet free. The trapped droplet is then transported through the water by moving the transducer.

2:35

3pPA6. Development of heavy metal ions detector driven by surface acoustic waves. Yue Liu, Chaohui Wang, and TengFei Zheng (School of Mech. Eng., Xi'an Jiaotong Univ., No. 28, Xianning West Rd., Xi'an 710049, Shaanxi, China , 492865529@qq.com)

Environmental pollution caused by heavy metals is a global problem. Most of the metal ions lead to serious health concerns by producing free radicals. Therefore, rapid and accurate detection of metal ions has become an urgent problem. We develop a highly sensitive detector to detect heavy metal ions driven by surface acoustic waves (SAWs). The detector consists of a micro-reaction cell, a three-electrode system for electrochemical measurements, and two focused interdigital transducers (FIDTs) that produce SAWs. The SAWs promote liquid mixing effectively by sufficiently stirring the aqueous solution; therefore, the oxidation current in electrochemical reactions is enhanced. Furthermore, the center angles of the FIDT and the distance between two FIDTs turn out to be critical parameters in reaction efficiencies. These two factors can affect the SAW radiation areas which can influence the reaction efficiencies. This study realizes rapidly and accurately detecting heavy metal ions and provide a basis for designing SAWs-based detector for different chemical reactions.

2:50

3pPA7. Standing surface acoustic waves controlling crystal shapes: The case of urea. TengFei Zheng, Yue Liu, and Chaohui Wang (School of Mech. Eng., Xi'an JIiaotong Univ., No. 28, Xianning West Rd., Xi'an 710049, Shaanxi, China, 492865529@qq.com)

It is important to control crystal shapes, for this is particularly significant in nano-technology and in medicine. In this paper, we use standing surface acoustic waves (SSAWs) based technique to control the urea crystal shapes in a microdroplet. The surface acoustic waves are excited by a sinusoidal electric potential applied to inter-digital transducers on the surface of a X-propagating lithium niobate (LN) single-crystal piezoelectric substrate. A microdroplet is placed on the waves propagation path. SSAWs radiate into the droplet and influence the crystallization process when the droplet evaporates. First, we investigated the effect of acoustic intensity. Second, we controlled the shape of the crystal by changing the wave frequency. Finally, we controlled the shape of the crystal by changing the LN surface properties. Moreover, the streaming induced by SSAWs was observed by particle image velocimetry. Moreover, it is easier to control the wave frequency, acoustic intensity, and the surface properties than to control acoustic cavitation. This research will promote the application of SSAWs in controlling crystal forms.

3:05

3pPA8. Overview of acoustical levitation for volumetric visual displays. Carl Andersson and Jens Ahrens (Chalmers Univ. of Technol., Sven Hultins Gata 8a, Gothenburg 412 58, Sweden, carl.andersson@chalmers.se)

Acoustical levitation is an area with many applications ranging from medical drug delivery to micro-particle sorting. An application which has gained attention recently is the creation of volumetric displays by using small levitating objects. Advances in inexpensive ultrasonic phased arrays have increased the availability of dynamically controllable beamformers which enables the manipulation of the levitating objects in time and space. This allows for interpreting the levitating objects similarly to pixels in a normal display yet with an additional spatial dimension. Most implementations so far are based on the so-called Gor'kov formulation of radiation force. This formulation coupled with numerical optimization allows for calculation of the phases of the individual elements in the array. By exploiting symmetries in the solution, it is possible to impose an acoustic trap signature phase pattern onto simple focusing methods. Using off-the-shelf mid-air ultrasonic haptics systems to provide multiple focus points to which the phase patterns are applied allows for real-time control of multiple levitating objects. Current systems are limited to a handful of individually controllable objects so visualization is limited to abstract information. We present an overview of the state-of-the-art and discuss limitations and possibilities. [Work supported by Horizon 2020, No. 737087.]

Session 3pPPa

Psychological and Physiological Acoustics, Speech Communication, and Education in Acoustics: Diversity in Auditory Perception and Speech Communication

Kelly L. Whiteford, Cochair
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Dominique A. Bouavichith, Cochair
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Evelyn M. Hoglund, Cochair
Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43204

Chair's Introduction—1:30

Invited Papers

1:35

3pPPa1. Racial categorization and word identification: The influence of sex, race and regional dialect. Yolanda F. Holt (Commun. Sci. and Disord., East Carolina Univ., 300 Moye Bv 3310-X HSB, MS 668, Greenville, NC 27834, holty@ecu.edu) and Tessa Bent (Indiana Univ., Bloomington, IN)

The speech signal provides information on talker characteristics including socio-ethnic affiliation and racial identity. Regional variation, both similar and divergent from White American English, has been described in African American English. However, it is unknown if such regional dialect variation influences listeners' racial categorization or word identification accuracy. This work evaluated the influence of listeners' sex, race, and regional dialect on racial categorization and word identification for Black and White talkers from two dialect regions within North Carolina. Black and White listeners (n = 23) from eastern and central North Carolina participated. In the racial categorization task, listeners heard /hVd/ words produced by male and female Black and White talkers from eastern and western North Carolina. Listeners categorized the perceived talker race for each token as Black or White. In the word identification task, the same listeners matched the speech tokens from the same talkers to one of fourteen /hVd/ words. Results showed an effect of listener sex on word identification accuracy such that female listeners were more accurate than male listeners. No effect of listener race or regional dialect was observed for either task. Follow-up analyses will investigate the interaction between listener and talker sex, race, and regional dialect.

1:55

3pPPa2. The southern shift and regional identity in Appalachia. Paul E. Reed (Communicative Disord., Univ. of Alabama, 909 Welsh Humanities Bldg., University of South Carolina, Columbia, South Carolina 29208, reedpe@email.sc.edu)

The Southern Vowel Shift (SVS) historically occurred across the Southern U.S. (e.g., Labov *et al.*, 1972; Feagin, 1986). Several recent studies document the retreat of the SVS in the urban South (Fridland, 1999; Prichard, 2010; Dodsworth and Kohn, 2012). However, Irons (2007) found the SVS advancing in rural Kentucky. Thus, Fridland notes the SVS might not be a supra-regional norm, serving rather as an ecological distinction within the South (2012:187). As an ecological distinction, SVS features might reflect a speaker's orientation toward differing ecologies. Thus, a speaker with a rural orientation might use more SVS features than a speaker who is less rurally oriented. The present paper investigates the SVS in rural speech examining three features: monophthongization of /aI/, rotation of /E/ and /e/, and rotation of /I/ and /i/. Data come from sociolinguistic interviews, reading passages, and words lists from 25 speakers from rural northeast Tennessee. Additionally, every participant completed a Rootedness Metric, a psychometric survey that quantifies place-based orientation. Results indicate that speakers exhibit SVS features, with reversals of the relative front vowel positions and monophthongization of /aI/ in all positions (cf. Irons, 2007). However, more rooted speakers exhibited the most advanced SVS features. Thus, the central difference of SVS features may not merely be ecological, rather the speaker's relationship to the differing regional ecologies.

2:15

3pPPa3. Proactive neural processing of native and non-native speech. Fernando Llanos (Commun. Sci. and Disord., Univ. of Pittsburgh, 507 Edmond St., 6, Pittsburgh, PA 15224-2036, f.llanoslucas@gmail.com), Rachel Reetzke (Health, Univ. of California Davis, Austin, Texas), and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

The impact of attention and language-experience in neural speech processing is typically assessed using sentences, words, and syllables that, when presented in isolation, may not engage the cortical speech network as well as more realistic continuous speech. Here, we explore the neuromodulatory effects of attention and language experience using continuous speech. We recorded electroencephalographic responses from native speakers of English and late Chinese-English bilinguals while listening to a story recorded in English. The story was mixed with a tone sequence, and listeners were instructed to focus either on the speech (attended speech condition) or the tone sequence (ignored speech condition). We used the multivariate temporal response function and the accuracy of a machine-learning based brain-to-speech decoder to quantify differences in cortical entrainment and speech-sound category processing. Our analyses revealed a more robust, context-independent neural encoding of speech-sound categories when they were attended and native. Interestingly, while cortical entrainment to speech was also enhanced by attention, the enhancement was stronger among non-native speakers. Our results suggest that while listeners can manage attention to improve the neural parsing of continuous speech via cortical entrainment, the benefits of attention for speech-sound category processing can be attenuated when the sounds are not native.

2:35

3pPPa4. Turn up the volume: Speech perception in noise for bilingual listeners. Erika Skoe (Dept. of Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., U-1085, Storrs, CT 06105, erika.skoe@uconn.edu)

Bilinguals, compared to monolinguals, have been reported to have stronger neural representation of the fundamental frequency (F0) of speech, as measured by the frequency-following response (FFR). In monolinguals, stronger FFRs to F0 have been associated with better speech perception in noise (SPIN), suggesting that bilinguals should outperform monolinguals on SPIN tests. However, the opposite is the case: bilinguals generally underperform monolinguals. To explain such findings, Krizman *et al*. (2017), proposed that the bilingual brain might *turn up the volume* on the neural representation of sound to compensate for reduced SPIN. The current work considers this possibility using a combination of FFR and SPIN data. The Revised Speech in Noise test was administered at signal to noise ratios (SNRs) of 0 and 3 dB to young adult monolinguals (n = 17) and early bilinguals (n = 26). Unlike the monolinguals, the bilinguals showed a drop in performance when the SNR dropped. Within the bilingual group, poorer SPIN performance correlated with stronger neural responses to F0, suggesting that sensorineural areas are recruited to increase the neural gain of the acoustic representation in a manner that inversely correlates with speech comprehension. These findings give new insight into the brain-behavioral relationships for the neural encoding of sound.

2:55

3pPPa5. Effects of aging on voice emotion recognition in cochlear implant users and normally hearing adults listening to spectrally degraded speech. Shauntelle Cannon and Monita Chatterjee (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, Shauntelle.Cannon@boystown.org)

Voice emotion recognition declines with age in normally hearing (NH) adults. However, the effects of aging on voice emotion recognition in NH listeners with spectrally degraded stimuli are neither known nor the effects of aging on adult cochlear implant (CI) users. This study explored age-related effects on voice emotion recognition in NH adults listening to spectrally degraded speech and aimed to compare these changes to age-related changes in adult CI users' voice emotion recognition with unprocessed speech. Participants listened to 12 emotion-neutral sentences, each spoken with 5 emotions (happy, sad, scared, angry, and neutral) by a male and female talker. Participants identified which emotion they heard while listening to sentences that were either unprocessed or CI-simulated. Preliminary results indicate declines in overall percent correct scores and increased reaction time with both age and increasing spectral degradation in NH adults. Results also suggest age-related effects on percent correct scores and reaction times within the CI group alone. These results have important implications in the aging population of adults with NH and with CIs because limitations in the quality peer to peer interactions have been associated with a decrease in perceived quality of life. [Work supported by R01 DC014233 and P20 GM109023.]

Session 3pPPb

Psychological and Physiological Acoustics and Speech Communication: Context Effects in Speech Perception II (Poster Session)

Christian Stilp, Cochair
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Matthew Winn, Cochair
Speech & Hearing Sciences, University of Washington, 1417 NE 42nd St., Seattle, WA 98105

All posters will be on display, and all contributors will be at their posters from 1:30 p.m. to 3:30 p.m.

Contributed Papers

3pPPb1. Sentence perception in noise by hearing-aid users: Relations with syllable-constituent perception and the use of context. James D. Miller (Res., Commun. Disord. Technol., Inc., 3100 John Hinkle Pl, Ste. 107, Bloomington, IN 47408, jamdmill@indiana.edu), Charles S. Watson (Res., Commun. Disord. Technol., Inc., Bloomington, IN), Marjorie R. Leek (Res., VA Loma Linda Healthcare System, Loma Linda, CA), David J. Wark (Commun. Sci. and Disord., Univ. of Memphis, Memphis, TN), Pamela Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), Sandra Gordon-Salant (Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Masked sentence perception by hearing-aid users is strongly correlated with three variables: (1) the ability to hear phonetic details as estimated by the identification of syllable constituents in quiet or in noise; (2) the ability to use situational context that is extrinsic to the speech signal; and (3) the ability to use inherent context provided by the speech signal itself. These conclusions are supported by the performance of 57 hearing-aid users in the identification of 109 syllable constituents presented in a background of 12-talker babble and the identification of words in naturally spoken sentences presented in the same babble. A mathematical model is offered that allows calculation of an individual listener's sentence scores from estimates of context utilization and the ability to identify syllable constituents. When the identification accuracy of syllable constituents is greater than about 55%, individual differences in context utilization play a minor role in determining the sentence scores. As syllable constituent scores fall below 55%, individual differences in context utilization play an increasingly greater role in determining sentence scores. When a listener's syllable constituent score is above about 71% in quiet, the listeners score in quiet will above about 55% in noise. [Watson and Miller are shareholders in Communication Disorders Technology, Inc., and may profit from sales of the software used in this study.]

3pPPb2. The role of linguistic variability in the perception of voice cues. Floor Arts, Etienne Gaudrain, Terrin N. Tamati, and Deniz Baskent (Dept. of Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Hanzeplein 1, Groningen, The Netherlands, f.arts@umcg.nl)

Talkers' voices play an important role in speech perception. Through voices, we identify individual talkers and can facilitate speech communication in challenging conditions (e.g., cocktail party situations). While previous research has suggested that several linguistic factors broadly influence talker perception, how these factors influence perception of the individual voice cues remains unclear. The current study investigated the role of linguistic variability in voice cue perception, specifically fundamental frequency (F0) and vocal-tract length (VTL). Just-Noticeable-Differences (JNDs) were obtained using a 3AFC adaptive paradigm. Effects of word status (words, nonwords), word characteristics (lexical frequency, neighborhood density), and nonword characteristics (phonotactic probability, neighborhood density) were examined. Results demonstrated that voice cue perception was influenced by linguistic variability. While overall similar for words and nonwords, F0 and VTL JNDs were affected by phonological information in nonwords, i.e., phonotactic probability, but not on lexical information, i.e., lexical frequency and neighborhood density. However, VTL JNDs varied less across linguistic conditions than F0 JNDs, suggesting different processing mechanisms for these voice cues. These findings provide better insight into the interaction of voice and linguistic information, which may also improve our understanding of speech perception processes in populations with limitations in voice perception, such as cochlear-implant listeners.

3pPPb3. Interactions between auditory quality, linguistic context, short-term memory, and speech recognition. Adam K. Bosen (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, adam.bosen@boystown.org)

The ability to recognize speech depends on several factors, including the quality of the auditory input, linguistic context, and the listener's cognitive abilities. These factors correlate with speech recognition, but less consideration has been given to how these factors interact. Here, we present evidence from two studies indicating that interactions exist between quality, context, and short-term memory. In study one, we demonstrate that contextual expectations across stimulus sets can determine the relationship between auditory quality and short-term memory. Specifically, recall of lists of digits is not affected by noise-band vocoding, whereas vocoding impairs both item identification and recall for lists of single syllable words drawn from a large, untrained set. In study two, we demonstrate that correlations between digit list recall and PRESTO sentence recognition are strongest when auditory quality is poor, whereas correlations between digit and word list recall weaken with decreasing auditory quality. This finding suggests that auditory quality and semantic context moderate the relationship between memory and speech recognition. We conclude that incorporating experimental measures of auditory quality, short-term memory, and recognition of speech materials with different contexts will provide a clearer perspective on how these factors relate to speech recognition in listeners with hearing loss.

3pPPb4. Examining use of context in sentences: The recruitment of perceptual and cognitive-linguistic abilities in younger and older adults. Daniel Fogerty, Rachel E. Miller (Commun. Sci. and Disord., Univ. of South Carolina, 1224 Sumter St., Columbia, SC 29208, fogerty@sc.edu), Jayne B. Ahlstrom, and Judy R. Dubno (Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Auditory-verbal contextual information facilitates speech recognition. In everyday listening, perceptual and cognitive-linguistic abilities influence speech recognition, but the role of these abilities in contextual use remains unclear. To assess the use of context, younger and older adults listened to sentences in multitalker babble and identified the final words from either high- or low-predictability sentence frames; difference scores between the two conditions indicate use of context. We also assessed (1) speech processing abilities, including masking release, talker segregation, and auditory-verbal closure, (2) cognitive abilities, including memory, inhibition, and speed of processing, and (3) linguistic abilities, assessed through verbal fluency and vocabulary knowledge. Compared to younger adults, older adults had better contextual use and vocabulary knowledge but had poorer recognition of final words with increasing age. These measures, combined with thresholds and age, were entered as predictor variables to explain performance for high- and low-context sentences. Preliminary results indicate that vocabulary knowledge and auditory-verbal closure were associated with contextual use and final word recognition in low-context sentences, respectively. Final word recognition in high-context sentences involved a combination of abilities. Thus, recruitment of perceptual and cognitive-linguistic abilities depends, in part, on the availability of sentence context. [Work supported by NIH/NIDCD.]

3pPPb5. Earlier music biases subsequent musical instrument categorization. Joshua M. Lanning and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., University of Louisville, Louisville, KY 40292, jmlann01@louisville.edu)

Perception of sounds occurs in the context of surrounding sounds. When spectral properties differ between earlier (context) and later (target) sounds, categorization of later sounds becomes biased through *spectral contrast effects* (SCEs). Past research has shown SCEs to bias categorization of speech and music alike. Additionally, the magnitudes of SCEs are not all-or-none but vary continuously in both speech and music categorization. Recently, the natural spectral composition of (unfiltered) sentences biased speech categorization via SCEs (Stilp and Assgari, under review). Here, we tested whether natural (unfiltered) music would similarly bias categorization of French horn and tenor saxophone targets. Preceding contexts were either solo performances of the French horn or tenor saxophone (unfiltered) or a string quintet processed to emphasize frequencies in the horn or saxophone (filtered). Categorization was influenced by SCEs in both unfiltered and filtered conditions, with more "saxophone" responses following horn / horn-like contexts and vice versa. Since unfiltered music produced SCEs in musical instrument categorization, this extends their influence to everyday listening conditions.

3pPPb6. Attention versus adaptation in processing talker variability in speech. Sung-Joo Lim, Jessica Tin, Allen Qu, and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Auditory Neurosci. Lab., Boston Univ., 610 Commonwealth Ave., Boston, MA 02215-2422, sungj.m.lim@gmail.com)

Speech processing is slower when listening to speech from multiple talkers compared to one continuous talker. Prior studies of talker variability mostly compared listening to long blocks of a continuous talker versus blocks where talkers constantly switch. It is thus unclear whether differences in processing efficiency are better understood as facilitation from perceptual adaptation to a talker or as interference from the attentional disruption caused by abrupt talker switches. It is also unclear how processing a single talker's speech becomes more efficient with ongoing exposure. Here, we examined how speech processing speed depends on preceding exposure to a talker. Listeners performed a speeded word identification task, in which they heard each talker's speech for 2–7 consecutive trials before the talker switched. Word identification was slowest when the talker switched and was expedited after a single exposure to a talker. However, additional exposure to a talker did not further improve word identification speed. More frequent talker switches in the preceding speech also led to slower subsequent word identification. Our findings suggest that speech processing efficiency does not depend on perceptual adaptation to the preceding talker; rather, slower speech processing after a change in talker reflects the costs of attentional reorientation.

3pPPb7. More information is more information: Working memory and visual context. Joanna H. Lowenstein and Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Rm. 2147, P.O. Box 100174, Gainesville, FL 32610, jlowenstein@phhp.ufl.edu)

Listeners with cochlear implants demonstrate diminished auditory-verbal working memory capacities, possibly due to a lack of durable codes in the memory buffer. Earlier studies suggest that the context provided by lip-read information should enhance those codes, with both the phonological form and dynamic nature of lip-read signals contributing to this facilitative effect. The purpose of this study was to test the hypothesis that lip-read signals would make uniquely beneficial contributions to the recognition of degraded speech. To test this hypothesis, three kinds of signals were used: unprocessed words, vocoded words, and nonspeech environmental sounds. Two kinds of visual enhancements were applied: (1) dynamic signals specifying the event that generated the signal or (2) pictures representing the object named or creating the signal. Eighty young adults with normal hearing were asked to recall order of eight-item lists in a closed-set format. All listeners heard lists in all three signal conditions (unprocessed, vocoded, environmental sounds) but half recalled order in each visual-enhancement condition. Adding lip-read information improved accuracy and eased cognitive demands for recall of vocoded words, but other visual information provided benefits as well, calling into question previous claims of the specialness of dynamic facial movements.

Session 3pSA

Structural Acoustics and Vibration, Engineering Acoustics, Noise, and Architectural Acoustics: Novel Damping Treatments

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

Benjamin Beck, Cochair
Applied Research Lab, Penn State University, The Pennsylvania State University, Applied Research Laboratory, P.O. Box 30, MS 300B, State College, PA 16804-0030

Hubert S. Hall, Cochair
Mechanical Engineering, The Catholic University of America, 9500 MacArthur Blvd., Bldg. 3, Rm. 252.05, Code 7310, West Bethesda, MD 20817

Invited Papers

1:00

3pSA1. Shaping shock transmission with lightweight elastomeric materials. Ryan L. Harne and Peter Vuyk (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., E540 Scott Lab, Columbus, OH 43210, harne.3@osu.edu)

Effective suppression of impulsive elastic waves requires the reduction of the transmitted shock pulse and the elongation of shock duration. Recent experimental studies with engineered, lightweight elastomeric materials suggest that these requirements are met to large extent. The materials capitalize upon the mesoscale geometry that is known to collapse in unique ways according to the internal geometric design and magnitude of the impact force. Yet, the relations among material design, collapse trend, and resulting shock mitigation remain unknown. This research seeks to shed light on the connections using digital image correlation techniques that uncover exact origins of energy distribution through mapping of local strain fields. With a sequence of controlled shock experiments, we first identify how the impact force magnitude governs the classification of shock mitigation capability of the materials. Then, the relative variations of such trends as tailored by the internal material geometry are examined. All together, the results illuminate the range of working conditions and material designs for which shock attenuation capability of the materials remains exceptional.

1:20

3pSA2. Broadband elastic black hole and vibration isolator based on the transformation method. Hexuan Gao and Zhihai Xiang (Dept. of Eng. Mech., Tsinghua Univ., N510b, Mengminwei Technol. Bldg., Beijing 100084, China, gaohx15@mails.tsinghua.edu.cn)

The transformation method has stimulated many interesting applications of manipulating electromagnetic and acoustic waves by using metamaterials, such as super-lens imaging and cloaking. These successes are mainly due to the form-invariant property of the Maxwell equations and acoustic equations. However, the similar progress in manipulating elastic waves is very slow, because the elastodynamic equations are not form-invariant. Here, we show that the expression of the elastodynamic potential energy can approximately retain its form under two restrictions: conformal mapping and using the material whose longitudinal wave velocity is much larger than the transverse wave velocity. Based on this finding, we use inhomogeneous isotropic material to design and fabricate an efficient 180-deg wave bender acting as vibration isolator, in which the incident elastic waves turn around and make little disturbance on the item being supported. In addition, an elastic black hole is designed in the same way, so that the elastic energy propagating near the black hole is mostly absorbed into the center and dissipated by the damping material.

Contributed Papers

1:40

3pSA3. Design optimization of three styles of acoustic black hole vibration absorbers. Cameron A. McCormick (Appl. Res. Lab, Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, cam634@psu.edu) and Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., State College, PA)

Structures whose thickness is tapered according to a power law exhibit the acoustic black hole (ABH) effect and can provide effective vibration absorption with a net reduction in mass. However, it remains unclear what constitutes the best design of ABH vibration absorbers, including how the power-law taper is implemented in practice. This talk will present a formal optimization study of three styles of one-dimensional ABH vibration absorbers. Each ABH is embedded in a simply supported beam, which is excited by a harmonic force at one end. A multi-objective approach is used to identify the set of ABH designs that optimally minimize the structure's vibration response and its overall mass. Results show that each style has a similar tradeoff between the two objectives but that the choice of how the

power-law taper is implemented can be significant. Finally, the optimal designs will be evaluated on other criteria that may be of importance in the practical implementation of such ABH vibration absorbers, including buckling load and sound radiation.

1:55

3pSA4. Experimental design for the accurate measurement of ultra-low damping of simple structures. Hubert S. Hall, James J. Dlubac, and Michael Kim (Signatures, Naval Surface Warfare Ctr. Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, hubert.hall@navy.mil)

As a means of validating numerical models, recent research has shown a need for methodology to measure the structural damping of very lightly damped structures (loss factor < 0.005) to a high level of accuracy. Traditional experimental methods of measuring structural damping must be altered to accurately capture the lightly damped response. Otherwise, inaccurate damping values will be calculated that are larger than those intrinsically found in the material/structure. This presentation focuses on experimental technique modifications required for frequency domain methods of measurement of ultra-low damped structures. First, the implications of less accurate capture of resonant peaks in frequency response functions will be explored. Typical short frame, uniform frequency spacing can result in large measurement errors with lightly damped test articles. Instead, long time histories, focused sine-based testing, zero-padding, and peak approximation methods should be utilized for improved accuracy. Additionally, the effects of isolation-related boundary conditions and instrumentation mass to measurement accuracy are explored and quantified.

2:10

3pSA5. Impact of viscoelastic damping properties on vibration response of scale models. Benjamin Beck (Appl. Res. Lab., Penn State Univ., The Penn State Univ., P.O. Box 30, MS 3200D, State College, PA 16804-0030, benbeck@psu.edu)

There are many industries that utilized scale models to test the behavior of systems before building full-size structures, such as aircraft, spacecraft, and marine vehicle design. While vibration and acoustic response may not be the only quantities of interest when testing scale models, scale models can be utilized to predict full-scale acoustic behavior. When damping treatments are to be applied to full-scale systems, there will also be a need to apply these treatments to the scale models. Typically, these damping treatments use viscoelastic materials due to their high loss factor. However, since the stiffness and loss factor of viscoelastic materials change significantly with frequency, viscoelastic damping treatments will not behave consistently between model- and full-scale. This work shows the impact of using the same viscoelastic damping treatment on scale models as designed for full-scale systems. As a test case, a model of a thin vibrating plate will have a constrained layer damper applied on one surface. The total surface velocity response of a full-scale and model-scale finite element plate with a viscoelastic constrained layer damper is compared. Methods of designing viscoelastic materials for scale-models will also be presented.

2:25

3pSA6. Laser Doppler vibrometer and bending wave based dynamic material characterization of organic aerogel panels. Max Miller (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, millem23@rpi.edu), Sadeq Malakooti (Dept. of Mech. Eng., Univ. of Texas at Dallas, Richardson, TX), Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), Nicholas Leventis (Dept. of Chemistry, Missouri Univ. of Sci. and Technol., Rolla, MO), and Hongbing Lu (Dept. of Mech. Eng., Univ. of Texas at Dallas, Richardson, TX)

To what extent aerogels are useful in noise and vibration control applications remains an open question. With the advent of low cost, facile synthesis, ductile, and purely polymeric aerogels, the prospects are alluring. Within the polyurea family of aerogels, multiple nanomorphologies exist with some being amenable to rheological paradigms. An inverse problem approach is applied to tease out material parameters with the aid of viscoelastic models and a pair of dynamic characterization techniques. A specialized transfer function driven, quasi-longitudinal wave eliciting, material characterization setup is refined and augmented with laser Doppler vibrometry. This effort seeks to improve repeatability and increase frequency limits, two factors whose absences plague many schemas. Intriguing properties previously uncovered with this method, such as broadband negative dynamic mass, are reexamined. In tandem, a bending wave excitation arrangement and characterization method is introduced. Low bending and quasi-longitudinal phase speeds in addition to high static and low dynamic moduli shed light on the enhanced noise and vibration control performance reported in the literature. This paper discusses the dynamic measurement method and possible applications which may benefit from this unique combination of properties.

2:40

3pSA7. Analysis of increased damping in arrays of attached resonators. John A. Sterling, Joseph F. Vignola (Mech. Eng., Catholic Univ. of America, 620 Michigan Ave., Washington, DC 20064, jsterling@gmail.com), Teresa J. Ryan (Mech. Eng., East Carolina Univ., Greenville, NC), and Sai T. Paruchuri (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

Research has shown that arrays of small dynamic attachments on a larger, primary structure can be tuned to significantly alter the time and frequency response of the system. Such attachments can be used to increase the apparent damping of a primary structure by transferring energy into the substructure and dissipating the energy there. By selecting the properties of the attachments to obtain specific distributions of element mass and mode frequency, the response of the primary structure can be altered to obtain specific spectral outcomes. Unfortunately, small errors in the distributions of mass and stiffness of the individual attachments have been shown to produce a significant degradation of intended performance. Until recently, metals were the practical option that was easily manufactured to the tolerances required. New three-dimensional-printing advancements have made plastics available with sufficient tolerances. A serendipitous discovery was that the increased damping shown in the plastics decreased the system's sensitivity to error. Previous analysis assumed low damping in the attached resonators. This work will show analysis with moderate to high damping levels for the attachments. Here, we will show experimental results using different plastic materials with moderate dimensional error.

Session 3pSC

Speech Communication: Second Language Speakers and Listeners (Poster Session)

Susannah V. Levi, Chair
Communicative Sciences and Disorders, New York University, 665 Broadway, 9th floor, New York, NY 10012

All posters will be on display, and all authors will be at their posters from 1:30 p.m. to 3:30 p.m.

Contributed Papers

3pSC1. A generational acoustic analysis of Mexican Spanish and English vowels. Edwin D. Reyes Herrera (Linguist, Macalester College, 1600 Grand Ave., Saint Paul, MN 55105, ereyeshe@macalester.edu)

Multiple studies have suggested that the Spanish vowel system is stable, with little to no variation (Delattre, 1996; Quillis and Esgueva, 1983), while others have also noted variation in monolingual Spanish (Boyd-Bowman, 1952; Canellada and Vicente, 1960; Delforge, 2008; Lipski, 1990; Lope-Blanch, 1964; Marín Galvés, 1995; Matluck, 1952; Quilis, 1999; Quilis and Esgueva, 1983) and bilingual Spanish in the United States (Fought, 1999; Godinez and Maddieson, 1985; Konopka and Pierrehumbert, 2008; Konopka, 2011; Ronquest, 2013; Willis, 2005). This study analyzes the vowel quality and duration of Spanish and English vowels by Mexican and Mexican-American speakers across different sociolinguistic generations in the United States (Silva-Corvalan, 1994), which has been absent from previous literature. An examination of the different generations is needed to account for the differing levels of Spanish and English use in the United States, as well as to examine how production of speech may change across generations. Results indicate all generations exhibit significant unstressed vowel reduction, though there is variation across the different groups. The amount of centralization and reduction also increases for each subsequent generation. In addition, Spanish influence is evident in the production of both G1 and G3 groups, though there is also evidence for acquired English speech.

3pSC2. A comparison of training effects on non-native tone sandhi production between American English and Cantonese speakers. Bei LI (The Hong Kong Polytechnic Univ., AG511, Hung Hom, Hong Kong, 199122libei@gmail.com), Yike Yang, Si Chen (The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), and Yunjuan He (Univ. of North Georgia, Hong Kong, Hong Kong)

Previous studies suggest that productions of Mandarin tone sandhi by both American English speakers and Cantonese speakers were perceived more native-like after a laboratory perceptual training, whereas little is known about the effects of tonal or non-tonal backgrounds. Ten Cantonese-speaking trainees and ten American English-speaking trainees matched in age and Mandarin proficiency were recruited to the pre- and post-training recording sessions. Elicited with audio and visual stimuli, participants naturally produced disyllabic real and wug words where the two Mandarin tone sandhi rules (T3 + T1/T2/T4 sandhi and T3 + T3 sandhi rules) should be applied. In total, 7680 sandhi syllables obtained from two sessions were perceptually evaluated by two phonetically trained Mandarin-speaking raters on a 101-point scale. Statistical results indicated that native tonal/non-tonal backgrounds influence Mandarin learners' improvement in the two sandhi rules differently. The Cantonese trainees outperformed the English trainees in the sandhi of T3 + T1/T2/T4 before training, and the two groups had statistically comparable performance after training, although both groups exhibited significant improvement. For the sandhi in T3 + T3, improvement occurred for the Cantonese trainees while not for the American trainees after training, suggesting that the successful learning of phonological T3 sandhi rule may require a tonal background.

3pSC3. Cross-linguistic pitch differences in bilingual speakers. Andrew Cheng (Dept. of Linguist,Univ. of California, Berkeley, Berkeley, CA 94720, andrewcheng@berkeley.edu)

Studies in cross-language pitch differences show that languages can fundamentally differ in average pitch range even when speaker differences are accounted for (e.g., Mennen *et al*., 2012). Such studies usually compare similar populations of monolingual speakers but not bilingual speakers of the languages in question. Bilingual development research, in turn, does not address average language pitch but rather segmental properties such as vowel formants. The current study fills the gap by examining the fundamental frequency (f0, or pitch) of both languages of English-Korean bilinguals. Twenty Americans of Korean descent (female = 13, average age = 24) recorded natural speech during a bilingual interview. Their speech was digitally recorded and transcribed, and f0 was analyzed using the inverse filter control method (Ueda *et al*., 2007). Average f0 per word, per language, and per subject was then calculated and compared across individuals and groups. Results showed that Korean has significantly higher f0 than English, regardless of gender or age. The implications for research in acoustics and speech communication include a greater consideration for bilingualism or bilingual modes of speech as a variable, as well as an open line of inquiry into further study of cross-language differences within the bilingual speaker, including as a result of code-switching.

3pSC4. Effects of perceptual training in Mandarin tone sandhi production. Si Chen (The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Shuwen Chen (The Chinese Univ. of Hong Kong, Hong Kong, Hong Kong), Yunjuan He (Univ. of North Georgia, Hong Kong, Hong Kong), Bei LI (The Hong Kong Polytechnic Univ., AG511, Hung Hom, Hong Kong, 199122libei@gmail.com), Yike Yang (The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), and Ratree Wayland (Univ. of Florida, Gainesville, FL)

It is well-established that the production of non-native lexical tone poses a great challenge to adult L2 learners (Hao,2012; Chang and Yao, 2016; Mok *et al*., 2018). The production of tonal patterns on disyllabic words is more challenging for non-native speakers because additional computational and/or lexical mechanisms are involved in correctly applying the tone sandhi rules in languages like Mandarin (Chen *et al*., 2017). Previous speech training studies have shown that perceptual training could improve both perception and production of non-native tones in isolation (Wang *et al*., 1999, 2003; Wayland and Li, 2008). The current study aims to further examine if perceptual training promotes learning of tonal patterns on disyllabic words by examining the production of two Mandarin Tone sandhi rules—the third tone sandhi and half-third tone sandhi by Cantonese learners. Native Cantonese speakers were trained with an identification task and a same/different discrimination task with both real and wug words. Their pre- and post-

training production was compared with native speakers' production, and it showed that perceptual training can lead to more successful learning of Mandarin tone sandhi rules based on acoustic and statistical analyses of tonal contours.

3pSC5. Acoustic characteristics of voiceless English fricatives (/f θ s/) produced by native English and native Japanese speakers. Katsura Aoyama (Univ. of North Texas, 1155 Union Circle #305010, Denton, TX 76703, katsuraaoyama@gmail.com), James E. Flege (Univ. of Alabama at Birmingham, Tuscania, Italy), Reiko Akahane-Yamada (ATR Int., Seika cho, Kyoto, Japan), and Tsuneo Yamada (Open Univ., Chiba, Japan)

This study examined productions of three English voiceless fricatives (/f θ s/) produced by native Japanese (NJ) and native English (NE) adults and children (16 participants each in 4 groups). The purpose of this study was to investigate acoustic characteristics of these fricative productions that were evaluated using intelligibility ratings in Aoyama *et al*. (2008). The following acoustic parameters were selected based on their importance in differentiating English fricatives (Jongman *et al*., 2000): noise duration, spectral moments, normalized amplitude, and relative amplitude. A total of 768 tokens were acoustically analyzed (256 each of /f θ s/). The results showed that there are many differences in acoustic characteristics between NJ and NE speakers. First, the durations of /θ/ were longer in the NJ speakers' productions than in the NE speakers' productions on average. Second, the spectral mean for /s/ was lower in the NJ speakers' productions than NE speakers' productions. Third, the NJ speakers' productions of /s/ did not differ from /f/ and /θ/ in normalized amplitude, whereas the NE speakers' productions of /s/ showed higher amplitude than /f/ and /θ/. These findings will be discussed and compared to the findings based on intelligibility ratings in Aoyama *et al*. (2008).

3pSC6. Pitch range, intensity, and vocal fry in non-native and native English focus intonation. Alex Hong-lun Yeung, Hyunah Baek, Chikako Takahashi, Joseph Duncan, Sharon Benedett (Dept. of Linguist, Stony Brook Univ., Stony Brook, NY 11794-4376, chikako.takahashi@stonybrook.edu), Jiwon Hwang (Asian & Asian American Studies, Stony Brook Univ., Stony Brook, NY), and Ellen Broselow (Linguist, Stony Brook Univ., Stony Brook, NY)

23 native English speakers (ES) and 25 native Mandarin speakers (MS) participated in a study of the production of contrastive focus prosody in English. The participants completed an interactive game in which they directed experimenters to decorate objects, producing sentences containing contrasting noun phrases in which either the adjective or noun was contrasted (e.g., Andy wants an orange diamond on his towel and a NAVY diamond/orange OVAL on Mindy's towel). Time-normalized average pitch and intensity contours extracted from a subset of the speakers suggest that while both groups distinguish adjective from noun focus, the MSs show a wider pitch range but smaller intensity drop than the ESs, consistent with a previously reported study of contrastive focus production (Takahashi *et al*., 2017). A surprising pattern in the data was that the MSs actually showed a stronger use of pitch cues on the focused noun than the ESs, which may have reflected the fact that many of the ESs exhibited creakiness toward the end of the sentence, restricting their use of pitch to mark focus on nouns. We argue that these divergent patterns reflect a combination of Mandarin L1 influence and innovative vocal fry prosody in native English speakers.

3pSC7. Changes in the phonetics of German language learners over a semester. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com) and Lauren Elliott (Psych., Carthage College, Kenosha, WI)

We compare the acoustic phonetics of German language learners at the beginning and end of a college semester. Participants recorded three utterances of five example sentences in the first and last months of the semester. Results from the early recordings documented that language learners were intermediate between native speakers and naive speakers in at least two different acoustic contexts. The voice onset time (VOT) of the /f/ sound was short in native speakers (more /f/-like) and long in naive speakers (more /v/-like). Similarly, the vowel space traversed during the vowel sound of "viel" had a smaller area in native speakers than in naive speakers. We predict that, on average, language learners will show change in both their VOT and their vowel space, becoming more similar to native speakers, while naive speakers will not show any change. We will use paired t-tests to compare the early and late productions from each speaker. We will also expand the array of acoustic features that we will examine to include additional vowel and consonant sounds.

3pSC8. Interplay of native and non-native vowels in Japanese late learners of English. Chikako Takahashi (Dept. of Linguist, SUNY at Stony Brook, Stony Brook, NY 11794-4376, chikako.takahashi@stonybrook.edu)

This study investigates perception of native (/i/-/e/) and non-native (/i/-/ɪ/) contrasts by Japanese late learners of English (N = 40). We hypothesized that speakers with greater proficiency in L2 might show more effects of L2 learning on L1 perception than speakers with lower L2 proficiency. We found first that self-rated proficiency correlated relatively well with L2 vowel contrast categorization ($r = 0.622$) indicating that late bilinguals can achieve more nativelike categorization of L2 vowels (/i/-/ɪ/) as their overall proficiency improves. Turning to the effect of L2 proficiency on L1 perception, we found that the bilinguals who exhibited more nativelike English /i/-/ɪ/ categorization in their L2 were more likely to have a broader /i/ category when identifying /i/vs./e/ in Japanese (compared to monolingual Japanese controls). We argue that while L2 English learners may be improving in identifying English /i/vs./ɪ/, the non-native /ɪ/ is not fully differentiated from their Japanese /i/ category, possibly providing atypical exemplars that influence the perceptual boundary of /i/ in their L1. The complex interplay of L1 and L2 vowels will be further discussed by reference to production data for the learners and native speaker controls for Japanese and English.

3pSC9. Second langauge production of French mid and high vowels: An articulatory perspective. Madeleine Oakley (Linguist, Georgetown Univ., 3700 O St. NW, Washington, DC 20057, mo643@georgetown.edu)

This study uses Ultrasound Tongue Imaging and acoustic data to investigate the articulatory strategies used by L1 English L2 French learners to produce round vowels. It has been suggested that learners have more difficulty in producing L2 phones that are "similar" to L1 phones than L2 phones that are completely "new" because learners use L1 categories to produce L2 phones (Flege, 1982; Kamiyama and Vaissiere, 2009). However, this claim is based solely on acoustic data. To this end, the present study records learners' articulatory strategies using Ultrasound during production of French round vowels /y, u, ø, o/ compared to English /u, o/. 1 L1 French speaker and 6 L2 French learners were recorded producing wordlists in French and English, using ultrasound, video recordings of lip protrusion, and audio recordings. Results show that learners do not, in fact, use L1 articulatory strategies to produce L2 phones. Additionally, articulatory data show that learners still have difficulty producing target-like tongue positions for new phones, despite having target-like acoustic productions, which may suggest that non-native vowels have an acoustic rather than an articulatory target.

3pSC10. Non-native perception of noise-vocoded speech. Ewa Jacewicz, Robert A. Fox, and Joy Lee (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

Research has established that non-native speakers consistently underperform compared to native speakers in challenging listening environments such as in noisy backgrounds. Furthermore, when presented with variable talkers and regional accents, non-native listeners are unable to benefit from available phonetic details to the same extent as do native listeners [Jacewicz *et al*., *J. Acous. Soc. Am.* **144**, 1864 (2018)]. The current study further inquired into this non-native processing deficit using noise-vocoded speech. Noise-vocoding not only models cochlear implant processing but is also informative regarding perception of degraded speech in normal-hearing listeners. Proficient Korean-English bilinguals were asked to identify talker dialect (Ohio, North Carolina) and talker sex responding to utterances processed through a noise-source vocoder at 4, 8, 12, and 16 channels. A separate task assessed intelligibility of this noise-vocoded speech. Since cues to

talker dialect are found primarily in fine spectral structure, we expected them to perform increasingly worse than native listeners with the decreasing number of channels. Both native American English controls and Korean-English listeners adhered to the predicted pattern across tasks and conditions. The study provides further evidence that the ability to utilize partial spectral information is compromised in non-native speech processing.

3pSC11. Effects of pitch contour and speaking rate on perception of foreign-accented speech. Rebecca F. Davis (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, rebecca.davis.1@louisville.edu), Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Eugene, OR), and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Perception of foreign accent is typically studied using an accentedness rating task. For example, native English listeners rate the degree of accentedness in sentences produced by non-native English speakers. However, in the past studies, it has been unclear that on what criteria participants used to judge accentedness. Here, native English speakers rated the accentedness of Korean-accented English sentences on a scale from 1 (strong accent) to 9 (little to no accent). Participants rated sentences that were unmodified or had one acoustic property removed. In one block, pitch contours of sentences were flattened and set to their mean values. In another block, speaking rates were set to the grand mean of all speaking rates (3.8 syllables/second). This way, changes in accentedness ratings across unmodified and modified sentences were attributable to the acoustic property that was removed. Accentedness ratings were not systematically influenced by manipulations of pitch contours but were influenced by speaking rate manipulations. Increasing the speaking rate (to 3.8 syllables/second) made sentences sound less accented than their unmodified versions; decreasing the speaking rate made sentences sound more accented than their unmodified versions. Results suggest that the speaking rate directly contributes to ratings of foreign accentedness.

3pSC12. Task-evoked pupillary response to completely intelligible accented speech. Drew J. McLaughlin (Psychol. & Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, drewjmclaughlin@wustl.edu) and Kristin Van Engen (Psychol. & Brain Sci., Washington Univ. in St. Louis, Saint Louis, MO)

Listening to second language- (L2-) accented speech is often described as an effortful process, even when L2 speakers are highly proficient. This increase in listening effort is likely caused by systematic segmental and suprasegmental deviations from native-speaker norms, which require additional cognitive resources to process (Van Engen and Peelle, 2014). In this view of speech perception, even when an L2 speaker is completely intelligible (i.e., their words can all be correctly identified), perception nonetheless requires increased cognitive load compared to native speech. We used pupillometry (the measure of pupil diameter over time) as a psychophysiological index of cognitive load to address this hypothesis. Task-evoked pupillary response (TEPR) was measured while participants listened to sentences produced by a Mandarin Chinese-accented English speaker and a standard American-accented English speaker. Results from a first experiment showed that TEPR was larger for L2-accented speech than the native speech, indicating greater cognitive load during speech processing. In a second experiment, we controlled for differences in the speech rate between the two speakers. Preliminary evidence from this replication also shows larger TEPR for the L2-accented speech condition. Together, this evidence suggests that processing accented speech—even when completely intelligible—requires additional cognitive resources.

3pSC13. Production and perception of two English vowels by advanced Colombian Spanish learners of English. Jenna Conklin, Olga Dmitrieva (Linguist, Purdue Univ., 610 Purdue Mall, West Lafayette, IN 47907, jconkli@purdue.edu), and Gina M. Pineda Mora (Universidad Nacional de Colombia, Bogota, Colombia)

This study investigates the acquisition of the English vowels [æ] and [ʌ] by Colombian learners of American English. Thirty speakers of Colombian Spanish residing in the United States participated in this study. A multiple-choice identification task demonstrated that participants were proficient at identifying [æ] and [ʌ], though the accuracy rate was somewhat lower than for a number of other English vowels used for comparison. When misidentified, [æ] and [ʌ] were most frequently confused with each other or with the low back vowel [ɑ]. In a production task completed by both bilingual Spanish speakers and monolingual English speakers, learners' [æ] and [ʌ] were distinct from native monolingual English productions. The nature of the acoustic differences between native speakers' vowels and learners' vowels suggested that Colombian speakers' English renditions of [æ] and [ʌ] were affected by their Spanish [a]. Further analysis revealed that learners' [æ] was not statistically different from their own production of Spanish [a]. These results are compatible with the assumption emerging from previous research that speakers of Spanish acquiring English assimilate, to a varying degree, English low and low-mid vowels to Spanish [a]—a single vowel found in this area of the vowel space in Spanish.

3pSC14. Changes to perceptual assimilation following training. Julia Kharlamenko, Heather M. Kabakoff (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, jk5096@nyu.edu), Erika Levy (Commun. Sci. and Disord., Columbia Univ., New York, NY), and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Adult speakers of American-English have difficulty perceiving front-back rounding contrasts in vowels, such as French /œ/-/o/. These difficulties stem in part from how learners map sounds in a second language onto native-language sound categories. We used a perceptual-assimilation task to test how perceptual training changes listeners' mapping of nonnative speech sounds. One listener group was trained with a bimodal distribution of stimuli drawn from an /œ/-/o/ acoustic continuum, a second was trained with a unimodal distribution, and a third (control) group completed no training. The training incorporated active learning with feedback and lexical support. In the assimilation task, listeners heard French vowels and were asked to select which of eight English /hVd/ words best matched the French stimulus. Results revealed general perceptual training effects and differences in both trained groups. For /o/, both trained groups differed significantly from untrained controls. However, for /œ/, listeners in the unimodal group did not differ from the untrained controls, but the bimodal group differed from the untrained controls. Thus, exposure to a unimodal distribution can alter perception, but a bimodal distribution may have stronger effects on perception. Bimodal distribution may be better for facilitating assimilation of nonnative sounds to native-like categories.

3pSC15. A comparison between native and non-native speech for automatic speech recognition. Seongjin Park and John Culnan (Dept. of Linguist, University of Arizona, Box 210025, Tucson, AZ 85721, seongjinpark@email.arizona.edu)

This study investigates differences in sentence and story production between native and non-native speakers of English for use with a system of Automatic Speech Recognition (ASR). Previous studies have shown that production errors by non-native speakers of English include misproduced segments (Flege, 1995), longer pause duration (Anderson-Hsieh and Venkatagiri, 1994), abnormal pause location within clauses (Kang, 2010), and non-reduction of function words (Jang, 2009). The present study uses phonemically balanced sentences from TIMIT (Garofolo *et al*., 1993) and a story to provide an additional comparison of the differences in production by native and non-native speakers of English. Consistent with previous research, preliminary results suggest that non-native speakers of English fail to produce flaps and reduced vowels, insert or delete segments, engage in more self-correction, and place pauses in different locations from native speakers. Non-native English speakers furthermore produce different patterns of intonation from native speakers and produce errors indicative of transfer from their L1 phonology, such as coda deletion and vowel epenthesis. Native speaker productions also contained errors, the majority of which were content-related. These results indicate that difficulties posed by English ASR systems in recognizing non-native speech are due largely to the heterogeneity of non-native production.

3pSC16. The development of perception of English tense-lax vowel contrast by monolingual English-speaking and bilingual Mandarin-English children. Lan Yu (Indiana Univ. Bloomington, 1020 E. Kirkwood Ave., Ballantine 821, Bloomington, IN 47405, ly29@indiana.edu)

This study explored the perceptual weighting utilized by children and adults for the perception of the English tense-lax vowel contrast in both the first language (L1) and second language (L2). Both the desensitization hypothesis (Bohn, 2000), which posits a universal bias toward duration cues on non-native vowel contrasts, and developmental perceptual weighting shift hypothesis in L1 (Nittrouer *et al*., 1993), which suggests a shift from dynamic to static cues developmentally, were tested in this study. Listeners were 4 English monolingual children (EC) with a mean age of 6:9, 4 Mandarin-English bilingual children (MEC) with a mean age of 8:4, 4 English-speaking adults (EA), and 4 Mandarin-English bilingual adults (MEA). In an identification experiment, listeners identified stimuli from *beat*/*bit* and *sheep*/*ship* continua differing in six acoustically equal duration and spectral steps. EC, MEC, and EA rely on spectral differences exclusively, yet MEA utilize an additional cue, duration, to identify this contrast. Findings suggest that desensitization needs to be interpreted with caution since other factors might greatly shift the perceptual weighting such as the age of L2 acquisition and the amount of L2 exposure. As for a developmental perceptual weighting shift, it only applies to the perception of consonants not vowels.

3pSC17. Non-native perception and first language experience: The case of English final stop perception by Saudi Arabic listeners. Sarah Alamri (GMU, 4400 University Dr., Fairfax, VA 22030, salamri4@gmu.edu)

This study investigated how listeners' native language [Saudi Arabic (SA)] could affect the perception of non-native (English) sound patterns. SA stopsare generally released, while English stops show free variation: released or unreleased. In a phoneme detection task, SA listeners were asked to detect English stops in coda position, and their accuracy and reaction time (RT) were measured. The influence of three factors of the final coda (i.e., release, voicing, and place of articulation) on the accuracy and RT was statistically analyzed using mixed effect models. The results indicated a significant interaction among the three factors on the listeners' accuracy but not on the RT. The results revealed that listeners relied mostly on release bursts as a phonetic cue to accurately detect the target phoneme. The significant interaction between release and voicing suggested that the unreleased voiced stops were easier to detect than the voiceless counterparts. Despite the difference in the accuracy, the listeners' RT was not significantly influenced by release, voicing, or place of articulation. The results suggested that L1 experience effects non-native perception if the target phoneme does not have enough acoustic information. Listeners can accurately detect unfamiliar sound patterns when the signal is acoustically rich.

3pSC18. Formant discrimination of speech and non-speech sounds in temporally modulated noise: Effect of language experience. Mingshuang Li, Can Xu, Chang Liu (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX 78712, limingshuang@utexas.edu), and Sha Tao (State Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Beijing, China)

Recent studies showed that the residency experience of 1–3 year in English-speaking countries could improve temporal dip listening against energetic masking of noise (i.e., masking release), resulting in better performances in identification of vowel, consonant, and sentence in temporally fluctuating noise. The current study aimed to explore whether this advantage would be extended to speech and non-speech perception discrimination. Three groups of listeners were recruited: English-native listeners (EN), Chinese-native listeners in US with long US residency experience (i.e., 2.5–4 years; CNUL), and Chinese-native listeners in the US with short US residency experience (i.e., less than half a year; CNUS). Thresholds of spectral discrimination of speech and non-speech stimuli were measured. EN listeners are expected to have the lowest thresholds and highest masking release in vowel formant discrimination, followed by CNUL listeners, with highest thresholds and lowest masking release for CNUS listeners. Furthermore, the current study will also examine whether the effect of language experience on masking release would be presented for non-speech spectral discrimination in noise. [Work supported by the China National Natural Science Foundation (31628009).]

3pSC19. Acoustic analysis of nasal and lateral consonants: The merger in Eastern Min. Ruoqian Cheng and Allard Jongman (Dept. of Linguist, Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, rqcheng@ku.edu)

The contrast between word-initial [n] and [l] is disappearing in many Chinese languages, including Eastern Min. Before investigating the status of the merger, we need to identify the acoustic cues that distinguish [n] from [l]. In English and Mandarin, languages with the [n] and [l] contrast, we examined: (1) Duration: consonant duration and consonant-vowel transition duration; (2) Formant frequencies: F1, F2 and F3 at the midpoint of the consonant; (3) Formant intensities: I1, I2, and I3 at the midpoint of the consonant; and (4) Relative amplitude (the amplitude difference between the consonant and the following vowel). Preliminary results show that [n] is significantly longer than [l] in Mandarin and English but not in Eastern Min. F2 of [n] is higher than F2 of [l] in English and Mandarin, whereas the direction is reversed in Eastern Min. Finally, the difference in relative amplitude between [n] and [l] is greater in English and Mandarin than in Eastern Min. Together, these results suggest a merger in progress in Eastern Min . Moreover, older Eastern Min speakers showed the merger to a greater degree than younger speakers, presumably because the older speakers use Mandarin less frequently.

Plenary Session and Awards Ceremony

Lily M. Wang,
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

Megan S. Ballard – For contributions to shallow water propagation and geoacoustic inversion

Woojae Seong – For contributions to geoacoustic inversion and ocean signal processing

Lori J. Leibold – For contributions to our understanding of auditory development

Robert W. Pyle, Jr. – For contributions to the understanding of the acoustics of brass musical instruments

Rajka Smiljanic – For contributions to cross-language speech acoustics and perception

Edward J. Walsh – For contributions to auditory physiology, animal bioacoustics, and public policy

Introduction of Award Recipients and Presentation of Awards

Rossing Prize in Acoustics Education to Stanley A. Chin-Bing

William and Christine Hartmann Prize in Auditory Neuroscience to Glenis R. Long

Distinguished Service Citation to David Feit

R. Bruce Lindsay Award to Adam Maxwell

Silver Medal in Engineering Acoustics to Thomas B. Gabrielson

Helmholtz-Rayleigh Interdisciplinary Silver Medal to Barbara G. Shinn-Cunningham

Gold Medal to William J. Cavanaugh

Vice President's Gavel to Scott D. Sommerfeldt

President's Tuning Fork to Lily M. Wang

Session 3eED

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

L. Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Ste. 300, Melville, NY 11747

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

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

Chair's Introduction—5:00

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will meet starting at 4:30 p.m.

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

Committees meeting on Tuesday are as follows:

Engineering Acoustics (4:30 p.m.)	McCreary
Signal Processing in Acoustics (4:30 p.m.)	Beckham
Acoustical Oceanography	McCreary
Animal Bioacoustics	Clements
Architectural Acoustics	French
Musical Acoustics	Breathitt
Physical Acoustics	Jones
Psychological and Physiological Acoustics	Carroll Ford
Structural Acoustics and Vibration	Stopher

Committees meeting on Wednesday

Biomedical Acoustics	Nunn

Committees meeting on Thursday

Computational Acoustics (4:30 p.m.)	Clements
Noise	Segell
Speech Communication	Carroll Ford
Underwater Acoustics	McCreary

Distinguished Service Citation

David Feit

2019

The Distinguished Service Citation is awarded to a present or former member of the Society in recognition of outstanding service to the Society.

PREVIOUS RECIPIENTS

Laurence Batchelder	1972	William J. Cavanaugh	1994
Robert W. Young	1973	John C. Burgess	1996
Betty H. Goodfriend	1973	Alice H. Suter	1997
Gerald J. Franz	1974	Elaine Moran	1999
Robert T. Beyer	1978	John V. Bouyoucos	2000
Henning E. von Gierke	1978	F. Avril Brenig	2000
R. Bruce Lindsay	1981	Thomas D. Rossing	2006
William S. Cramer	1984	Charles E. Schmid	2008
Stanley L. Ehrlich	1986	Uwe J. Hansen	2011
Samuel F. Lybarger	1986	Richard Stern	2011
Frederick E. White	1987	Allan D. Pierce	2015
Daniel W. Martin	1989	Paul D. Schomer	2015
Murray Strasberg	1990	Susan B. Blaeser	2016

ENCOMIUM FOR DAVID FEIT

. . . for service and contributions to the Acoustical Society of America, especially as Treasurer

LOUISVILLE, KENTUCKY • 15 MAY 2019

David Feit has a long history of service to national defense, to the study of acoustics and to the Acoustical Society of America (ASA). David is known in the acoustics community for his mastery of acoustics, and his wonderful writing and teaching skills.

David began his studies in engineering at Columbia University, receiving four degrees and culminating in an Engineering Science Doctorate in 1964. As a graduate student, he worked at the Stevens Institute of Technology Experimental Towing Tank, which began his long career in working with ships. After graduation, David worked at Cambridge Acoustical Associates (CAA), Inc., and then joined in 1973 the Carderock Division as the Head of the Vibrations and Acoustics Technology Division of the Ship Acoustics Department. He also served as the Liaison Scientist for Acoustics and Mechanics at the Office of Naval Research European Office from 1988 to 1990. David's service to the Navy was so exemplary that he received a letter of appreciation from the Commander of the Naval Sea Systems Command and the Navy's Meritorious Civilian Service Award in 1982.

Not only did David serve the Navy as a valued researcher, he has also served the acoustics academic community as an Adjunct Professor at Catholic University, George Washington University, Naval Academy, and University of Maryland for varying periods. David also lectured at the Massachusetts Institute of Technology (MIT) on the "Fundamentals of Ship Acoustics" in the summers of 1982 to 1986 and from 1993 to 1998. David is an author of 31 papers, 17 of which were published in the *Journal of the Acoustical Society of America*. He was recognized for his considerable achievements with the Trent-Crede Medal from the ASA in 1999, and the Per Bruel Gold Medal from the American Society of Mechanical Engineers in 2003.

David's most lasting contribution to the acoustics community is his book, *Sound, Structures and Their Interaction*, which was coauthored with Miguel C. Junger. This book is a superbly well-written, seminal work and has had an immeasurable impact on our community. It can be a challenge to find a superfluous sentence in either this book or in any of David's numerous technical papers and presentations, which are all clear, concise, and focused. (David also has a reputation for using equations on napkins as common form of communication sometimes to the dismay of family and restaurant waiters.) These qualities spill over to David's technical discussions with colleagues and, when combined with his modest and unassuming manner, earned him a (closet) sobriquet. Rather than by his initials DF, David was known as EF, mimicking the commercial tag line popular in the 1970's, "When EF Hutton talks, people listen".

Lastly, David has been an exemplar for service to our society. He joined the Society in 1965 and was elected a Fellow in 1975. David began participating in ASA meetings as a presenter and session organizer in the 1960s, and served as Technical Program Chair for the spring 1995 meeting held in Washington, DC. David has been a member of the Technical Committee on Structural Acoustics and Vibration (TCSA) since 1965, serving as its chair from 1992 to 1998. As the TCSA chair he was also a member of the Technical Council.

David has served as associate editor of the *Journal of the Acoustical Society of America* since 2005. In 1993, he and Miguel Junger donated the copyright of *Sound, Structures and Their Interaction* to the ASA so it could be included in its classic works series.

However, David's most important and significant role in the Society has been as the ASA treasurer since 2000. David managed ASA's 5 million dollar annual budget and provided guidance to the Executive Council on the Society's financial affairs. As treasurer, he served as *ex officio* on seven committees related to ASA's financial endeavors. He has tirelessly worked with other officers, committee chairs, staff members, and the ASA accountants and auditors to insure that the ASA's finances are handled efficiently and accurately. During my

term (M.I.) as president, I observed, firsthand, David's commitment to the financial health of the ASA.

David's reputation is worldwide. Whether from his Columbia University days, his stints at CAA and the David Taylor Research Center, or his European assignment, David has colleagues, acquaintances and friends everywhere.

David's long standing devotion and service to the acoustics community and the ASA, make this award very well deserved, and it could not be going to a finer person. And, as we all know, "When DF Feit talks, people listen".

MARCIA ISAKSON
JOEL GARRELICK
SUSAN FOX
JAMES LYNCH

R. BRUCE LINDSAY AWARD

Adam Maxwell

2019

The R. Bruce Lindsay Award (formerly the Biennial Award) is presented in the Spring to a member of the Society who is under 35 years of age on 1 January of the year of the Award and who, during a period of two or more years immediately preceding the award, has been active in the affairs of the Society and has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics, or both. The award was presented biennially until 1986. It is now an annual award.

PREVIOUS RECIPIENTS

Richard H. Bolt	1942	Joseph M. Cuschieri	1991
Leo L. Beranek	1944	Anthony A. Atchley	1992
Vincent Salmon	1946	Michael D. Collins	1993
Isadore Rudnick	1948	Robert P. Carlyon	1994
J. C. R. Licklider	1950	Beverly A. Wright	1995
Osman K. Mawardi	1952	Victor W. Sparrow	1996
Uno Ingard	1954	D. Keith Wilson	1997
Ernest Yeager	1956	Robert L. Clark	1998
Ira J. Hirsh	1956	Paul E. Barbone	1999
Bruce P. Bogert	1958	Robin O. Cleveland	2000
Ira Dyer	1960	Andrew J. Oxenham	2001
Alan Powell	1962	James J. Finneran	2002
Tony F. W. Embleton	1964	Thomas J. Royston	2002
David M. Green	1966	Dani Byrd	2003
Emmanuel P. Papadakis	1968	Michael R. Bailey	2004
Logan E. Hargrove	1970	Lily M. Wang	2005
Robert D. Finch	1972	Purnima Ratilal	2006
Lawrence R. Rabiner	1974	Dorian S. Houser	2007
Robert E. Apfel	1976	Tyrone M. Porter	2008
Henry E. Bass	1978	Kelly J. Benoit-Bird	2009
Peter H. Rogers	1980	Kent L. Gee	2010
Ralph N. Baer	1982	Karim G. Sabra	2011
Peter N. Mikhalevsky	1984	Constantin-C. Coussios	2012
William E. Cooper	1986	Eleanor P. J. Stride	2013
Ilene J. Busch-Vishniac	1987	Matthew J. Goupell	2014
Gilles A. Daigle	1988	Matthew W. Urban	2015
Mark F. Hamilton	1989	Megan S. Ballard	2016
Thomas J. Hofler	1990	Bradley E. Treeby	2017
Yves H. Berthelot	1991	Yun Jing	2018

CITATION FOR ADAM D. MAXWELL

. . . for contributions to the understanding and application of therapeutic ultrasound

LOUISVILLE, KENTUCKY • 13 MAY 2019

Adam Maxwell grew up in the Seattle area in a family entrenched in the sciences. His grandfather was an aerospace engineer for Boeing and his father a development engineer at Philips/ATL. Adam followed suit and enrolled in electrical engineering at the University of Washington with a focus on circuits, devices and technology. His introduction to research came during his sophomore year when a family friend with connections to the Applied Physics Laboratory encountered Adam ringing-up groceries as his part-time job. She thought the kid's talents might be put to better use. Bob Odom, then at the APL, steered Adam to Mike Bailey's lithotripsy research lab. Adam was impressive from the very start and it wasn't long before he was building custom equipment and repairing the lithotripter high-voltage supply. It is noteworthy that as an undergraduate Adam led the way in developing an inexpensive PVDF membrane hydrophone that proved to be an accurate and reliable alternative to the fiber-optic probe hydrophone used to measure lithotripter shock waves. Adam built dozens of these hydrophones, designed and made preamplifiers for them, tested sensitivity and modeled their frequency response. His design was subsequently licensed by U-Washington to a Seattle company that successfully marketed the device. Before he left for graduate school Adam also played a key role in a seminal research study demonstrating the action of shear waves as a mechanism for stone fracture in shockwave lithotripsy.

At the University of Michigan, Adam was awarded an NSF fellowship to pursue his PhD in Biomedical Engineering under the supervision of Zhen Xu. Here, Adam found his niche in research on acoustic cavitation. In this work Adam made key contributions to the understanding of bubble cloud formation in histotripsy, the mechanical disruption of tissue by short pulses of high amplitude ultrasound. His work provided the first evidence that peak positive pressure contributes to the cavitation cloud responsible for histotripsy erosion and that the high frequency content of the shock wave was significant in reflecting and inverting the pressure from the initial bubble. Adam discovered he could create histotripsy with acoustic pulses having predominant negative pressures and leveraged this observation to pursue an intrinsic cavitation threshold for fluids and tissue-like materials. This led to the demonstration that histotripsy erosion depended on tissue type and that certain cancers could be eroded while leaving otherwise healthy tissue and blood vessels intact. Adam also developed histotripsy techniques to disrupt blood clots and discovered that histotripsy could be used to make dense collagenous connective tissue more compliant, a finding he would build on years later to demonstrate proof of principle in softening fibrous scars to recover tissue flexibility in Peyronie's disease. Working with Tim Hall, one of the co-inventors of histotripsy, Adam helped develop 3D printable transducers that are now ubiquitous in histotripsy and lithotripsy research. His deep understanding of histotripsy and his expertise with the instrumentation to produce it made him a valuable consultant to HistoSonics, a company spun out by U-Michigan to commercialize histotripsy. In addition, Adam investigated small-scale interactions between solitary bubbles and cells leading to the generation of acoustic streaming to trap small particles. Overall, Adam's studies combined aspects of shock formation, radiation force and acoustic streaming to the understanding of how ultrasound at extreme pressure can generate and grow a cavitation cloud and how the cloud erodes tissue and tissue-like materials.

At the completion of his graduate studies, Adam was attracted back to U-Washington by the opportunity to contribute to a rapidly growing partnership between the APL, the Center of Industrial and Medical Ultrasound and the Department of Urology. He turned his attention to applying cavitation to the treatment of stone disease. Adam built upon his mechanistic understanding of stone breakage to invent and develop burst wave lithotripsy (BWL), a direct competitor to conventional shock wave lithotripsy. Within 5 years

of Adam's invention, BWL is the key technology of a spin-off company (SonoMotion) that is already testing BWL in human patients and is working toward FDA regulatory clearance to market a device. This technology has also been integrated into the combined diagnostic and therapeutic ultrasound system NASA is developing for space exploration.

Adam is a builder intent on translating new technologies into practice. He chooses challenging, significant problems and finds answers through well-designed and carefully conducted research. His aim has been to develop and refine innovative ultrasound devices that will improve treatment outcomes for patients over a spectrum of clinical indications and applications. Adam has co-authored over 35 issued or pending patents covering many of his discoveries. In his young career he has already played an essential role in establishing foundational technologies for three biotech and medical device companies, HistoSonics for the non-invasive, non-thermal ablation of cancerous tumors; Matchstick Technologies for cavitation-based fragmentation of DNA in multi-well assay plates; and SonoMotion for the non-invasive fragmentation of urinary stones using BWL. Adam is also developing an acoustic tractor beam with which he has successfully trapped, lifted and carried urinary stones through the pig bladder. Such a technology holds unique potential for use in clearing stones from the kidney, non-invasively capturing and steering them through the complex three-dimensional path of the urinary space.

Adam has long been active in service to the ASA. He is a highly engaged member of the Biomedical Acoustics Technical Committee, has chaired several ASA special sessions and has authored two *Acoustics Today* articles, one on lithotripsy and one on histrotripsy. He reviews manuscripts regularly for a dozen journals, including enough JASA papers to be on the editors' thank you list. Those who know Adam appreciate not only his capabilities as a researcher and critical thinker but view him fondly and with great respect for his willingness to work hard to help others succeed. Adam is exceptionally creative, but he is also selfless, genuine and generous.

On behalf of those who have had the privilege of working with Adam we welcome this opportunity to celebrate his many accomplishments. Adam D. Maxwell is in all respects a colleague highly deserving of the 2019 R. Bruce Lindsay Award.

MICHAEL R. BAILEY
J. BRIAN FOWLKES
JAMES A McATEER

Silver Medal in Engineering Acoustics

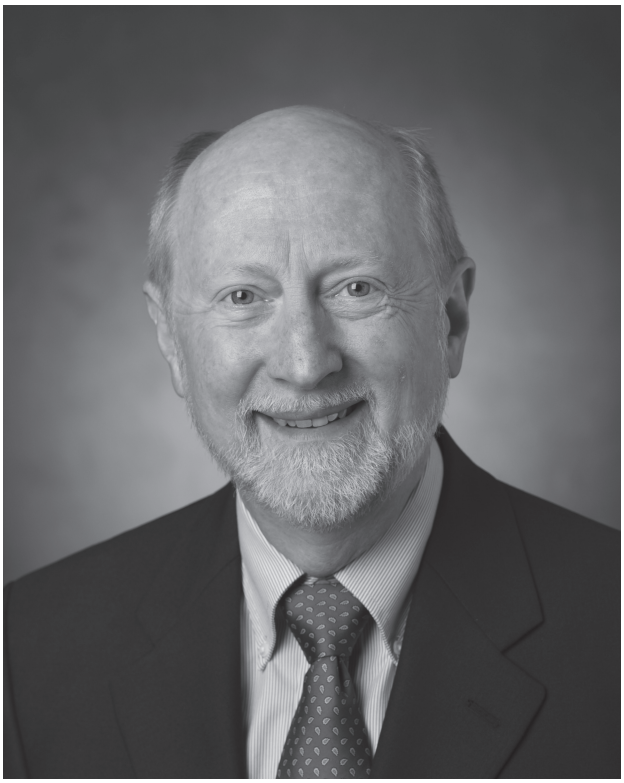
Thomas B. Gabrielson

2019

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

Harry F. Olson	1974	James E. West	1995
Hugh S. Knowles	1976	Richard H. Lyon	1998
Benjamin Bauer	1978	Ilene J. Busch-Vishniac	2001
Per Vilhelm Briiel	1982	John V. Bouyoucos	2004
Vincent Salmon	1984	Allan J. Zuckerwar	2007
Albert G. Bodine	1986	Gary W. Elko	2011
Joshua E. Greenspon	1989	John L. Butler	2015
Alan Powell	1992		



CITATION FOR THOMAS B. GABRIELSON

. . . for contributions to the understanding of novel transducers and their intrinsic limitations imposed by thermal and quantum physics

LOUISVILLE, KENTUCKY • 11 MAY 2019

More than most members in the Acoustical Society of America (ASA), Tom Gabrielson is perceived to be someone who works in the same field as yours. If you study shallow water propagation and use sonobuoys, you think that must be Tom's specialty. If your work involves micromachined sensors, you think that's his specialty. Same story for atmospheric propagation and infrasound. For the correct perspective, you only need to see Tom as a fully dedicated scientist and experimentalist. Whether the project is sponsored research or just a source for his own personal amusement, Tom brings the same innovative talent and attention to detail. One independent project of this type was a seismic sensing system installed at his Pennsylvania home that was capable of detecting waves crashing on the New Jersey shore some seventy miles away.

Tom received a B.S.E.E. from the New Jersey Institute of Technology and M.S. and Ph.D. degrees from Pennsylvania State University. He is also a Licensed Professional Engineer in Pennsylvania and is Board Certified by the Institute of Noise Control Engineering. Tom began his professional career in 1974 at the Naval Air Development Center (NADC) in Warminster, PA. While completing his Ph.D., he spent a year as a Visiting Professor at the Naval Postgraduate School in Monterey, CA, and then returned to NADC. At NADC, he developed several models for the prediction of acoustic propagation underwater, and led the Naval Air Warfare Center Harsh Environments Program to measure acoustic propagation under environmentally complex conditions. As a flight-rated project specialist, he collected acoustic data around the globe in P-3 aircraft. While at NADC, Tom was the recipient of both their Scientific Achievement Award and Outstanding Independent Research Project Award. He was also elected a Fellow of ASA in 1995. After NADC closed in 1996, Penn State was fortunate to hire Tom into the Graduate Program in Acoustics and the Applied Research Laboratory where he currently serves as Professor of Acoustics and Senior Scientist.

Tom's scientific interests have always been eclectic, ranging from astronomy and geosciences to many acoustical sub-disciplines. Constant throughout has been Tom's enjoyment of the design, fabrication, and testing (i.e., engineering) of complete instrumentation systems, ranging from the raw sensor and its housing, through the signal-conditioning electronics, to the calibration and subsequent signal-processing at the end of any transduction chain. In addition to this lab-focused effort, he also has an addiction to field experiments that exploit his custom transducer systems. Tom's field experiments have included measurement of underwater sound propagation in shallow water, military jet noise studies, and acoustic measurement of solid transport by mountain streams. His field experiment that received the most public attention was his work with a team from Penn State to locate survivors after the World Trade Center collapse using remotely deployable microphones and accelerometers connected to signal-conditioning electronics and recording devices built into a backpack. More recently, his measurements of the noise produced by natural-gas compressor stations was featured on National Public Radio.

Tom is best known in the Engineering Acoustics community for his contributions to the development, calibration, test, and analysis of both conventional and unconventional transducers, as well as conventional transducers used in unconventional contexts, such as the geophones used as underwater acoustic particle velocity detectors. He has investigated thermoacoustic engines as underwater sound sources, electron tunneling junctions as miniature accelerometers, and accelerometer-based acoustic intensity sensors.

Tom has been the author or coauthor of papers in the *IEEE Transactions on Electronic Devices*, ASME *Journal of Vibration and Acoustics, AIAA Journal, US Navy Journal of*

Underwater Acoustics, and the *Journal of the Acoustical Society of America*. He holds four patents and has been an invited speaker at numerous meetings and workshops sponsored by the Office of Naval Research, American Society of Mechanical Engineers, Institute of Noise Control Engineering, American Geophysical Union, American Vacuum Society, and ASA.

Perhaps his broadest impact on our contemporary understanding of transduction is based on his careful analysis of the fundamental limitation imposed by physical noise sources to the signal-to-noise ratio of micro-electro-mechanical sensors (MEMSs) that are fabricated using the technology developed for large-scale integrated circuits. A measure of that influence is the number of citations (417), patent references (35), and electronic downloads (2,694) of his now classic paper entitled "Mechanical-thermal noise in micro-machined acoustic and vibration sensors" (*IEEE Transactons on Electronic Devices* 40(5), May 1993).

Over the last several years, Tom's focus has shifted to the measurement and analysis of atmospheric infrasound. Again, it is the quality of the novel transducers, suppression of wind noise, and such transducers' absolute calibration in the lab and *in situ* calibrations in the field that has become the latest beneficiary of his unique combination of attention to the transduction mechanism and the use of innovative acoustical pre-filtering, signal-conditioning electronics, and subsequent signal processing that has become the hallmark of his unified approach. Tom has also explored the scientific and historical context of those measurements, having written articles on the 1883 eruption of Krakatoa and the history of atmospheric acoustic propagation for *Acoustics Today*.

Although ASA awards a separate prize for acoustics education, this encomium would not be complete if it did not recognize Tom's contributions to Engineering Acoustics through the two graduate courses he has offered each year in Penn State's Graduate Program in Acoustics, and many other short courses. Anyone who has ever had the pleasure of attending one of Tom's presentations can appreciate the unique insight, clarity, originality, and wry humor he brings to such talks. His students' enthusiasm for his lectures, and particularly his lecture demonstrations and homework assignments that involve field experiments, is legendary. He is the only faculty member we know, who has ever received a standing ovation at the end of his course and has received such appreciation in classes over the span now of 30 years.

Tom has been an invaluable asset to the community of Engineering Acoustics, and to the entire Acoustical Society, for nearly forty years. It is fitting that his name should be added to the list of Silver Medal recipients "*for contributions to the understanding of novel transducers and their intrinsic limitations imposed by thermal and quantum physics*."

STEVEN L. GARRETT
DAVID L. GARDNER
JAMES F. McEARCHERN
ARTHUR W. HORBACH
TIMOTHY M. MARSTON

Helmholtz-Rayleigh Interdisciplinary Silver Medal
in
Psychological and Physiological Acoustics, Speech Communication, and Architectural Acoustics

Barbara G. Shinn-Cunningham

2019

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler	1997	James V. Candy	2008
David E. Weston	1998	Ronald A. Roy	2010
Jens P. Blauert	1999	James E. Barger	2011
Lawrence A. Crum	2000	Timothy J. Leighton	2013
William M. Hartmann	2001	Mark F. Hamilton	2014
Arthur B. Baggeroer	2002	Henry Cox	2015
David Lubman	2004	Armen Sarvazyan	2016
Gilles A. Daigle	2005	Blake S. Wilson	2017
Mathias Fink	2006	Kenneth S. Suslick	2018
Edwin L. Carstensen	2007		

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993

CITATION FOR BARBARA G. SHINN-CUNNINGHAM

. . . for contributions to understanding the perceptual, cognitive, and neural bases of speech perception in complex acoustic environments

LOUISVILLE, KENTUCKY • 13 MAY 2019

Barbara Shinn-Cunningham attended Brown University as an undergraduate, receiving her B.Sc. in Electrical Engineering in 1986. She then attended the Massachusetts Institute of Technology (MIT) for her M.S. and Ph.D. degrees in Electrical and Computer Engineering. At MIT, Barb worked with Nat Durlach, Pat Zurek, and others in the Research Lab of Electronics, studying spatial hearing. Her work focused on temporal and spatial interactions, including the precedence effect and adaptation to changes in received sounds due to factors such as reverberation, head orientation, and visual inputs. She used both psychoacoustic experiments and mathematical models to understand listeners' capabilities, including adaptation to displaced sources and receivers (heads). Barb received her Ph.D. in 1994 and became a mother shortly after graduation, having the first of her two boys, Nick and Will, with her husband Rob Cunningham. Barb pursued her family life along with her wide interests while she worked part-time as a post-doc, including time at the Sensimetrics Corporation in the Boston area. She returned to academia by joining Boston University (BU) in 1997 with a primary appointment in Cognitive and Neural Systems and a joint appointment in Biomedical Engineering.

Professor Shinn-Cunningham thrived at BU, where she became full Professor of Biomedical Engineering in 2008. In 2011, she established the Center for Computational Neuroscience and Neural Technology as its Founding Director. In 2016, she established the Center for Research in Sensory Communications and Neural Technology, again as Founding Director. Her work with these research centers brought together researchers and graduate students from several academic departments. These centers stimulated interactions among a broad cross section of researchers at BU and beyond. In 2018, Professor Shinn-Cunningham left BU to join Carnegie-Mellon University (CMU) as Director of its new Neuroscience Institute and was appointed Professor of Biomedical Engineering, Psychology, and Electrical and Computer Engineering.

Professor Shinn-Cunningham's research interests broadened over her career from early interest in electrical instrumentation, to hearing psychophysics, to the neurophysiological basis of hearing abilities, and to the development of an integrated understanding of the interactions of all of these aspects with the natural world of communication. Her interests have increasingly been drawn to speech communication, and to the processing of those acoustic signals that are of central importance to human culture. Professor Shinn-Cunningham uses a broad set of empirical and mathematical tools to address speech and hearing in complex situations, including listening tasks that involve multiple speech sources in complex environments with both auditory and visual inputs. She studied the dynamics of onsets and offsets of sources, effects of reverberation, and the effects of cross-sensory and *a priori* knowledge. Her listening tasks involve speech understanding and incorporate multiple aspects of perceptions such as attentional control and visual cues. She also broadened her work with bilateral stimulation in complex environments to include electroencephalography and brainstem potentials. She has extended her work and interests to include a broad cross-section of experimental subjects: normal-hearing listeners, listeners with multiple types of hearing losses (including "hidden hearing losses"), and listeners with more central impairments including challenges like autism and even traumatic brain injury from blast exposure. From a conceptual perspective, this work involves basic mechanisms of speech and hearing, including basic neural coding of waveforms, as well as complex processing and hypothesis testing that allows listeners to separate independent sources, even in environments with reflections and coupled visual inputs.

In her integrative approach to understanding and separating out these multiple effects, Professor Shinn-Cunningham has a strong record of ongoing grant support from multiple

agencies and organizations, already including the National Institute of Deafness and Other Communication Disorders, the National Science Foundation, the Office of Naval Research, and private foundations. Her publications are numerous, over 114 articles in peer-reviewed journals, 48 papers in conference proceedings, and 10 book chapters. Her citations approach ten thousand.

Professor Shinn-Cunningham has actively served the profession through participation in several organizations. Her service to the ASA began with election to the Technical Committee on Psychological and Physiological Acoustics in 2000 and continued with multiple administrative committees. She was elected to its Executive Council (2010-2013), and subsequently as Vice President (2013-2016). She currently serves on the ASA Finance Committee.

Along with her array of intellectual and technical skills, Professor Shinn-Cunnigham cares deeply about her students and their development as scientists. When she was named recipient of the ASA Student Council Mentor Award, the nominators were effusive in their praise of her enthusiasm and excitement for what she, and they, were able to accomplish both professionally and personally. Working with her, one quickly realizes how much she truly loves what she does and how infectious that zest can be. Her enthusiasm for new scientific directions, and her ability to think about questions large and small has helped guide some of the best of the next generation of scientists.

One of the most important aspects of her work, which is very much in the spirit of the Helmholtz-Rayleigh Award, is Professor Shinn-Cunningham's efforts in organizing sessions and conferences that allow discussions by people from a diverse group of research backgrounds and interests. She not only works across a span of research areas, but she also stimulates sharing, comparisons, and collaborations among researchers in different areas. This is reflected in the many special sessions that she has organized and often chaired, representing many environments and organizations (including ASA). This is a natural outgrowth of her grants and contributions across multiple fields, including psychophysical measurements and modeling, neurophysiological recordings and modeling, speech communication characterization both experimentally and conceptually, and the behavioral impact of architectural acoustics. Overall, Professor Shinn-Cunningham is a consummate collaborator. Some of these collaborations she no doubt initiated herself, but others are instances in which she was invited to participate, because of her wide experience and insight. She is a wonderful colleague for many, with diverse ideas, connections, and passions across fields.

In her personal activities, Barb shows a remarkable diversity. She plays the oboe and the English horn and was recently the featured soloist with the Concord Orchestra. She is also an avid fencer and enjoys traveling and the out of doors with her family. Somehow she fits all this into a life that is incredibly productive and that includes many people. She is a wonderful colleague for many, with diverse ideas, connections, and passions across fields.

It is entirely appropriate that the Acoustical Society of America presents the Helmholtz-Rayleigh Interdisciplinary Silver Medal to Barbara G. Shinn-Cunningham–a remarkable scientist who has had enormous influence within the psychoacoustic, physiological, and speech-communication communities as well as the wider neuroscience field.

H. STEVEN COLBURN
PEGGY B. NELSON
WILLIAM M. HARTMANN

Gold Medal

William J. Cavanaugh

2019

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

Wallace Waterfall	1954
Floyd A. Firestone	1955
Harvey Fletcher	1957
Edward C. Wente	1959
Georg von Békésy	1961
R. Bruce Lindsay	1963
Hallowell Davis	1965
Vern O. Knudsen	1967
Frederick V. Hunt	1969
Warren P. Mason	1971
Philip M. Morse	1973
Leo L. Beranek	1975
Raymond W. B. Stephens	1977
Richard H. Bolt	1979
Harry F. Olson	1981
Isadore Rudnick	1982
Martin Greenspan	1983
Robert T. Beyer	1984
Laurence Batchelder	1985
James L. Flanagan	1986
Cyril M. Harris	1987
Arthur H. Benade	1988
Richard K. Cook	1988
Lothar W. Cremer	1989
Eugen J. Skudrzyk	1990
Manfred R. Schroeder	1991
Ira J. Hirsh	1992
David T. Blackstock	1993
David M. Green	1994
Kenneth N. Stevens	1995
Ira Dyer	1996
K. Uno Ingard	1997
Floyd Dunn	1998
Henning E. von Gierke	1999
Murray Strasberg	2000
Herman Medwin	2001
Robert E. Apfel	2002
Tony F. W. Embleton	2002
Richard H. Lyon	2003
Chester M. McKinney	2004
Allan D. Pierce	2005
James E. West	2006
Katherine S. Harris	2007
Patricia K. Kuhl	2008
Thomas D. Rossing	2009
Jiri Tichy	2010
Eric E. Ungar	2011
William A. Kuperman	2012
Lawrence A. Crum	2013
Brian C. J. Moore	2014
Gerhard M. Sessler	2015
Whitlow W. L. Au	2016
William M. Hartmann	2017
William A. Yost	2018

CITATION FOR WILLIAM J. CAVANAUGH

". . . for practical applications to building design and education in architectural acoustics, and for service to the Society."

15 MAY 2019 • LOUISVILLE, KENTUCKY

Bill Cavanaugh has changed the world as we hear it. He has been at the forefront of a broad spectrum of research and consulting in architectural acoustics for over 60 years. His service to the Acoustical Society of America, to many related societies, and to the science and art of acoustics in general has been deep and pervasive. Although always eager to redirect credit to his colleagues, most of us know at least some of the roles he played that have helped to define the entire field of building acoustics.

William J. Cavanaugh and the Acoustical Society of America were both born in 1929– Bill in Boston. During World War II he attended The English High School, the first public high school in America, a prep school for students who had dreams of attending nearby Massachusetts Institute of Technology (MIT). Although Bill was too young to serve, during WWII, he told his father that he wanted to join the Army Corps of Engineers instead of entering the MIT freshman class in 1946. His father gave him that life changing advice only a father can give. "Bill why don't you just try it out for a year. If you don't like it, you can join the service." Bill entered MIT as a civil engineering major that summer where he met Walter Hill, an architecture student who asked Bill to help with a project which would be displayed in the Rotundra. This peaked his interest and Bill decided to switch his major to Architecture where none other than Robert Newman was one of his professors and mentors. Bill graduated in 1951 with a Bachelor in Architecture degree and was awarded the American Institute of Architects Student Medal. He received his commission as 2nd Lieutenant and was immediately ordered to active duty. where he served as a unit training and staff officer with the 6th Armored Division and U.S. Army Corps of Engineers, for the duration of the Korean Conflict. Shortly before the armistice was signed, Bill married Ginny Huff, who he met as a student, in 1953. He remained in the Army Reserves with the Army Corps of Engineers until 1982, retiring with the rank of full Colonel. He was awarded by the Army's Meritorious Service Medal in acknowledgement of his 31 ½ years of continuous service.

After his service, Bill joined the trailblazing acoustical consulting firm of Bolt Beranek and Newman (BBN) in February 1954, from which most acoustical consulting firms can trace a lineage, and where he was guided and deeply influenced by its famous partners, Dick Bolt, Leo Beranek and Robert Newman. Bob Newman, a remarkable and animated teacher, became Bill's immediate mentor, and encouraged Bill's own teaching, always emphasizing expressive language that would be persuasive to architects and engineers, and always with an irrepressible enthusiasm. Alongside his consulting practice, Bill soon began teaching acoustics classes at the MIT School of Architecture, the Boston Architectural College, the Rhode Island School of Design, Harvard School of Public Health and Rodger Williams College. Bill's inspirational teaching led many students to become future clients, and some even became acousticians. Bill once had Tony Hoover sit in on a few classes he taught at the Boston Architectural Centre and, to inspire his younger colleague said "OK, Tony you take over the class now."

Bill has been deeply involved in nearly all aspects of architectural acoustics and noise control consulting. He has consulted on thousands of projects of all building types, requiring skillful interaction with architects, engineers, and clients, as well as the public. He has a uniquely friendly, thorough, educational, and effective consulting style. Insights and research based on his projects have been the source of most of his contributions to the acoustical community at large. There are five areas of Bill's contributions to acoustics and its practical application that are worth special consideration; masking, outdoor venue sound propagation, cinema sound quality, the practice of architectural acoustics consulting, and teaching and mentoring.

Bill was the lead author on the seminal paper ''Speech Privacy in Buildings," J. Acoust. Soc. Am. 34, 475-492 (1962), that had the daring and the scientific evidence to suggest that adding appropriate background sound could improve acoustical privacy. More research and papers followed, leading to the entire industry of masking systems, and making open plan offices viable by increasing the potential for speech privacy, enhanced concentration, and decreased interruption. Speech privacy remains a high priority, especially, with new federal mandates for privacy in health-care facilities.

Bill's interest in the difficult problem of neighbors' complaints of sound from outdoor amphitheaters led to and has continued to influence the entire field of concert sound monitoring systems, associated methods for improved community relations, and development of acoustical criteria for outdoor concert venues. He has earned a reputation as a skilled negotiator, and sometimes peacemaker in difficult disputes.

Bill's work on cinemas during the 1980's and 1990's with so much activity in improving movie sonic quality led to extensive investigation and testing of movie theaters across the country, resulting in criteria that are now the foundation for quantifying appropriate sound isolation, HVAC noise, and finish treatments. This work had major influences on THX systems and certification of cinemas, which in turn have greatly affected the exploding home entertainment industry.

Bill's outreach to other acousticians and his optimistic vision of the future of acoustical consulting were instrumental in the formation, growth, and vigor of the National Council of Acoustical Consultants (NCAC) and the Institute of Noise Control Engineering (INCE). The benefits to consultants, architectural acousticians, and acoustics practice in general are ubiquitous, dramatic, and enduring. Bill served as President of the National Council of Acoustical Consultants (NCAC 1977-79), and was awarded their C. Paul Boner Medal for Distinguished Contributions to the Acoustical Consulting Profession in 1983, and the inaugural Laymon N. Miller Award for Excellence in Acoustical Consulting in 2015, awarded jointly by NCAC and INCE. He also served as President of the Institute of Noise Control Engineering (1993).

Bill's teaching and mentoring are of special interest. Bill is uniquely generous with his time and support to so many on various aspects of our own careers, research, and teaching. After 17 years with BBN, Bill established, Cavanaugh Tocci Associates, with ASA Fellow, Gregory Tocci., and is regularly represented at ASA, NCAC, and INCE meetings, and boasts three Past Presidents of the NCAC. He was a leader in establishing the Robert Bradford Newman Student Award Fund, a subcommittee of the ASA Technical Committee on Architectural Acoustics, which sponsors Newman Student Medal Awards for excellence in the study of architectural acoustics, as well as the Theodore J. Schultz Grants for excellence in teaching. Acoustics books and teaching aids have benefited from Bill's influence as well, including his essential role in originating the various ASA books based on poster sessions, including "Halls for Music Performance: Two Decades of Experience, 1962-1982," through "Halls for Music Performance: Another Two Decades of Experience, 1982-2002". Bill is the principal co-author of "Architectural Acoustics: Principles and Practice", published by Wiley (1st Ed. 1999, 2nd Ed. 2010). Bill's reputation has generated many invitations to write chapters, forewords, and articles on architectural acoustics and noise control, as well as deep participation in many Standards Working Groups.

Bill's service to the Society includes as three consecutive years as Member of the ASA Executive Council, member and Chair of the TCAA, member of the Technical Committee on Noise, and chair or member of numerous administrative committees including Ethics and Grievances, Regional Chapters, Public Relations, Rules and Governance, and more. In fact, Bill was the first Chair of the College of Fellows and of Archives and History. Bill served as Chair of the Wallace Clement Sabine Centennial Symposium in June 1994 and was responsible for the Historical Display at the Society's 50th Anniversary in 1979. He served on the 75th Anniversary Committee and was co-editor with Henry Bass of the ASA 75th Anniversary commemorative book. His work with local chapters and other organizations is outstanding. And, he was awarded the ASA Distinguished Service Citation in 1994, and the ASA Wallace Clement Sabine Award in December 2006.

For many years Bill has worked diligently, and often behind the scenes, for the benefit and recognition of others. He firmly believes, and his actions demonstrate, that much can be accomplished if no one cares who gets the credit. His wise and generous council has provided a bedrock foundation for the high standards for technical, personal, and professional conduct expected of practicing architectural acousticians. Bill's selfless outreach and mentoring to other consultants and acousticians has influenced many outstanding careers in acoustics.

Ginny, his wonderful, loving wife of 57 years, who passed peacefully in 2010, his five children (including Bill, Jr. who died in 1982 at age 24 while on active duty in the US Air Force), his grandchildren, and his great grandchildren are the loves of his life. He is also in love with acoustical consulting, and it shows.

His countless friends and admirers are proud to congratulate Bill on this richly-deserved ASA Gold Medal. Bill Cavanaugh, CONGRATULATIONS and THANK YOU!

K. ANTHONY HOOVER
LEO L. BERANEK
ALLAN D. PIERCE

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Session 4aAA

Architectural Acoustics, Signal Processing in Acoustics, and Noise: Methods and Techniques Used for Simulation of Room Acoustics

Bruce C. Olson, Cochair
AFMG Services North America LLC, 8717 Humboldt Avenue North, Brooklyn Park, MN 55444

Ana M. Jaramillo, Cochair
Ahnert Feistel Media Group, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444

Chair's Introduction—8:15

Invited Papers

8:20

4aAA1. Level of detail in room-acoustic simulation. Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

The quality of present-day room acoustic simulations depends on the quality of the boundary conditions and of the underlying CAD room models. A "high-resolution" room model does not mean that it needs to have a visually perfect geometrical fine structure. To our experience, the required resolution of objects or surfaces does not need to be higher than about 1 m. In this presentation, an auralization engine is briefly introduced which uses a set of models of the same room but with a graduated level of detail (LOD). These different models can account for more physical correctness especially for very low-frequency specular reflections. Furthermore, a good estimate of scattering coefficients is essential. The relevance of the uncertainty of scattering coefficient data is discussed in a review on perception tests with varied surface scattering. Finally, guidelines for creation of CAD models are proposed.

8:40

4aAA2. Should we still rely on statistical calculations for the prediction of reverberation time? Ana M. Jaramillo (AFMG Services North America, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu) and Bruce Olson (Olson Sound Design, LLC, Brooklyn Park, MN)

Based on the conditions for the use of the most commonly used reverberation time equations, we have created room examples in EASE to compare how they correlate with ray tracing predictions and established a guideline on when we can rely on simple statistical predictions. The results show that statistical predictions are not always accurate, and the differences do not always go in the same direction, making it impossible to simply account for the under/over-estimation of the method.

9:00

4aAA3. Modelling the effects of spectators on speech intelligibility in a typical soccer stadium. Ross Hammond (School of Mathematics, Computing and Electronics, Univ. of Derby, Derby, United Kingdom), Peter Mapp (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com), and Adam J. Hill (School of Mathematics, Computing and Electronics, Univ. of Derby, Derby, United Kingdom)

Public address system performance is frequently simulated using acoustic computer models to assess coverage and predict potential intelligibility. Simulations are most-often completed in unoccupied spaces as this provides worst-case scenario intelligibility due to the reduced absorption. When the typical 0.5 speech transmission index (STI) criterion cannot be achieved in voice alarm systems, due to design difficulties, justification must be made to allow contractual obligations to be met. An expected increase in STI with occupancy can be used as an explanation, though the associated increase in noise levels must also be considered. However, numerous approaches exist when modelling the people which can produce significant discrepancies. This work demonstrates typical changes in STI for different spectator conditions in a calibrated stadium computer model. This includes different audience modelling approaches, distribution, capacity, posture (standing/seated), and atmospheric conditions. The effects of ambient noise are also considered. The results can be used to approximate expected changes in STI caused by different spectator conditions.

9:20

4aAA4. Acoustics simulations to inform the designs of large worship and entertainment spaces to the client and contractor. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

Computer based acoustical simulations can quickly communicate important information in visual and audible formats that have a strong and immediate impact on the non-acoustician decision makers and designers of a project. Two large spaces were modeled ($>$6000 m^3), one renovation, one new construction; simulations were used to help better understand the consequences of different design approaches for amplified sound and acoustical design, as well as handing value engineering response in a timely manner. Attention is given to the modeling's role in helping to sort out the paradigms of perception of the project team and then to inform the design options of the clients and end users.

9:40

4aAA5. Use and misuse of auralization. Wolfgang Ahnert (Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

Auralization was developed as a tool in the 30s. The historic overview over this development starts by using scale models as a design tool which is used until now. Here, the needed components are explained, and the pros and cons will be discussed. With the use of computer simulation in the end of the 60s, the presentation of auralized files started around 1990 first considered as a toy. Today the use of auralization is widespread. This paper describes the development of the technical tools to present auralized signals available for binaural reproduction without and with head tracker and by using loudspeaker reproduction without and with crosstalk cancelation. Nowadays, Acoustic labs with Ambisonics reproduction or similar technologies are used. In this presentation, the advantages of auralization are named including all positive properties to demonstrate the achieved simulation results to different client groups. But also, the misuse of auralization is shown in detail by using found examples.

10:00–10:15 Break

10:15

4aAA6. New tools to auralize simulation results with EASE 5.0. Tobias Behrens (ADA Acoust. & Media Consultants GmbH, Arkonastr. 45-49, Berlin 13189, Germany, tbehrens@ada-amc.eu), Khaled Wazaefi (ADA Acoust. & Media Consultants GmbH, Berlin, Deutschland, Germany), and Wolfgang Ahnert (AFMG Ahnert Feistel Media Group, Berlin, Germany)

The new software package in EASE allows the production of binaural files to check the quality of the simulation results and to make these results audible for music or speech samples in real time. Also, tests with head trackers have been made. Since 5 years, the software allows generating B-format files of second order. To reproduce sound fields based on these 9 files, a sound lab has been built. This lab will be represented and explained. A post-processing software for EASE allows to reproduce not only the calculated simulation files but in comparison also measured files by using the microphone Ambeo VR. Additionally, VR glasses generate realistic 3D-visuals, in the same model as used for acoustic simulation. That way realized acoustic treatments in the room become visible and audio-visual representations are possible. Results for comparison between simulated rooms and measured real rooms will be discussed.

Contributed Papers

10:35

4aAA7. Room acoustic simulation as a means to affect a musical composition for a location specific performance. Edwin S. Skorski (Interior Architecture, Univ. of North Carolina - Greensboro, 102 Gatewood Studio Arts Bld., 527 Highland Ave., Greensboro, NC 48859, skors1es@cmich.edu) and Steven J. Landis (Music, Univ. of North Carolina - Greensboro, Greensboro, NC)

Computer model simulations of existing interior spaces are often generated to document and analyze room acoustic characteristics. In this case study, a large, multi-tiered public atrium is analyzed for its potential use as a performance space. Furthermore, the analysis is also used to transform an existing musical composition into a location specific performance piece. The computer simulation highlights acoustic characteristics believed to be good for musical performance as well as those considered defects. Taking into account the unique room acoustic qualities of this non-traditional performance space, an existing musical composition is rewritten resulting in a space-dependent arrangement. Musical variables transformed due to the analysis include tempo, pacing, register, as well as source and receiver positions. Of specific interest are the room characteristics typically considered acoustic defects which are purposefully exploited to strengthen the impact of the performed piece. These include non-optimal reverberation times, sound focusing, and echoes. Acoustic analysis of the room and recordings of the composition will be presented.

10:50

4aAA8. An analysis of ceiling geometry within active learning classrooms. Edwin S. Skorski (Interior Architecture, Univ. of North Carolina - Greensboro, 102 Gatewood Studio Arts Bld., 527 Highland Ave., Greensboro, NC 48859, skors1es@cmich.edu)

The architectural designs and furnishing of active learning classroom spaces are playing an increasingly important role in the facilitation of modern educational methods. Traditional static classroom spaces effectively support a lecture style of teaching where student participation is passive. Due to their rigid space plan, they are poor at encouraging interaction among students and teachers. Conversely, active learning spaces promote innovative teaching methods where quick room re-configuration allows for discussion groups of various sizes, the simultaneous use of a variety of teaching methods, and provides greater opportunity for the incorporation of technology into the classroom. From a room acoustic perspective, the increase in room arrangement flexibility leads to a complex acoustic environment where the spatial relationship between the source and the receiver is highly variable. This study uses digital modeling and computer simulation to analyze the effects of the ceiling geometry as it relates to the active learning classroom acoustic environment. Specifically, the study focuses on changes in speech intelligibility and reverberation time as the overhead plane is manipulated.

11:05

4aAA9. Evaluation of shape grammar-generated diffuser arrays. Timothy Hsu (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, 535 W. Michigan St., IT 371, Indianapolis, IN 46202, hsut@iu.edu) and Jonathan Dessi-Olive (Architecture, Georgia Inst. of Technol., Atlanta, GA)

This paper proposes a means of evaluating arrays of quadratic residue diffusers (QRDs) generated through a grammar-based generative design method. Design processes for architectural acoustics are often highly conventional: acoustical designers have preferred to use historical examples, known equations, and standard principles of performative success. This is particularly true for surface treatments using diffuser products that are aggregated in ways that perform sufficiently but are visually predictable and monolithic. In the first phase of this project, a shape grammar approach to design acoustic diffuser arrays was proposed as a means of addressing the issue of design homogeneity in architectural acoustics and to break current habits of uniform deployment of diffusion treatments in spaces. A set of shape rules were proposed that generate non-uniform and sometimes surprising arrays of QRDs. This paper aims to expand demonstrate phase two, which includes the following: (1) clarification and further development of shaper grammar rules, (2) proposal of initial methods to evaluate the acoustic performance of these arrays, and (3) calculation of quantitative metrics. Numerical simulations will show time and directivity responses for these shape grammar generated diffuser arrays. Furthermore, diffusion and scattering coefficients will be presented as well as other proposed evaluation metrics for these larger arrays.

11:20

4aAA10. Design of a multiple-slope sound energy decay system with string and block coupling methods. Xuhao Du (Dept. of Mech. Eng., Univ. of Western Australia, 35 Stirling Hwy., Perth, WA 6009, Australia, xuhao.du@uwa.edu.au), Jie Pan (Dept. of Mech. Eng., Univ. of Western Australia, Crawly, WA, Australia), and Andrew Guzzomi (Dept. of Mech. Eng., Univ. of Western Australia, Perth, WA, Australia)

Driven by the need of both speech intelligibility and music perception, multiple-slope sound decay in a room has been studied for decades. To step further, two different room coupling methods are proposed for achieving decay with quadruple, quintuple, or higher slope numbers. Starting from a triple-slope sound decay system, the understanding of energy flow in such a system is developed as well as the relationship between the decay turning points and the aperture size. Each decay slope represents the acoustic characteristic of the corresponding dominating room, and the dominating durations are determined by the couple condition. Based on the understanding of its energy flow, two different coupling methods, the string coupling and block coupling, are developed for achieving specific non-exponential sound decay with the specific number of slopes, slope values, and times of turning point. These are controlled by parameters including the room quantity, the reverberation of each room, and the coupling aperture sizes. Based on the above coupling methods, a few examples are simulated with the diffusion equation for verifying the energy flux and achieving a specific multiple-slope sound energy decay pattern inside the room.

THURSDAY MORNING, 16 MAY 2019 CLEMENTS, 8:30 A.M. TO 10:00 A.M.

Session 4aAB

Animal Bioacoustics: Marine Mammal Bioacoustics

Michael A. Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

4a THU. AM

Contributed Papers

8:30

4aAB1. Best available science? Are NOAA fisheries marine mammal exposure noise guidelines up to date? Michael A. Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

NOAA fisheries employs a set of in-water noise exposure guidelines that establish regulatory thresholds for ocean actions that impact marine mammals. These are established based on two impact criteria: level A—a physiological impact including "Permanent Threshold Shift" (PTS) and level B—a behavioral impact or disruption. Recently, the level A exposure thresholds were reconciled to the frequency-dependent hearing sensitivities of five classes of marine mammals based on the work done more than a decade ago (Southall *et al*., 2007). Since that time, much more work has been published on behavioral impacts of various noise exposures, and consideration of more variables such as frequency-dependent noise propagation characteristics, cumulative, concurrent, and continuous exposures, and noise impacts on marine soundscapes has entered into the discussion—but have not been incorporated into the NOAA Fisheries guidelines. Some of these variables will be highlighted, suggesting that it may be time to reevaluate the thresholds for level B exposures.

8:45

4aAB2. Detection and classification beaked whale vocalization calls based on unsupervised machine learning algorithm. Kun Li and Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, Broussard Hall, Rm. 103, 240 Hebrard Blvd., Lafayette, LA 70503, kxl1737@louisiana.edu)

Currently, passive acoustic monitoring (PAM) becomes a more useful tool to record and analyze cetacean's calls, such as beaked whales and sperm whales. To study the regional population variability of beaked whales in the northern Gulf of Mexico, the acoustic data were collected by the Littoral Acoustic Demonstration Center-Gulf Ecological Monitoring and Modeling (LADC-GEMM) consortium at three different sites in the vicinity of the 2010 oil spill site. One of the challenging goals is an identification of different species of beaked whales autonomously in large datasets. A multistage detector was developed to detect and classify the beaked whale species. The data processing results show that two major species of beaked whales (Cuvier's and Gervais') were acoustically observed in the study area. The results suggest that Cuvier's beaked whales tend to be more active at the deeper sites (about 1500 m) and Gervais' beaked whales prefer the shallower site which is about 1000-m deep. The results highlight the need

for high spatial resolution acoustic monitoring and support the ecological niche hypothesis. The results also show that the acoustic activity for two major species of beaked whale have distributed throughout the day, and beaked whales do not exhibit seasonal preference for the Mississippi Valley site. The important new insights into the population structure and habitat preferences of different species of beaked whales in the northern Gulf of Mexico were obtained.

9:00

4aAB3. Detection of dolphin burst-pulses off Cape Hatteras, North Carolina, correlated to oceanographic features. Stephen B. Lockhart, Mike Muglia, and Lindsay Dubbs (Univ. of North Carolina Coastal Studies Inst., 850 NC 345, Wanchese, NC 27981, sblockhart.z0@gmail.com)

To assess the ecological impact of extracting energy from the Gulf Stream, the University of North Carolina Coastal Studies Institute has deployed a mooring on the continental slope off Cape Hatteras at a depth of 230 m, equipped with an Acoustic Doppler Current Profiler, CTD, and a hydrophone. Analyzing 16 months of data, we automatically detected dolphin "quacks" or "barks", using two detectors. First, we used a pitch detector to automatically detect such signals over a specified range of pitch values. Next, we used a matched filter approach. All detections were reviewed manually to eliminate false alarms. For these signals, we found a strong correlation with temperature and salinity at the bottom; the vocalizations were detected when the water was relatively cooler and fresher. As the Gulf Stream meanders seaward of the mooring site, the temperature and salinity there both decrease. Since this cooler water is higher in nutrients, one explanation for the correlation is that the marine mammals are attracted to this more productive water. Alternatively, the meandering Gulf Stream may influence either (a) the *acoustic propagation* around the mooring and/or (b) the *acoustic noise* around the mooring. Evidence for each alternative will be presented.

9:15

4aAB4. Understanding detectability variations for density estimation of marine mammals. Thomas Guilment, Natalia Sidorovskaia, Kun Li (Dept. of Phys., Univ. of Louisiana at Lafayette, UL BOX 43680, Lafayette, LA 70504-3680, thomas.guilment1@louisiana.edu), and Christopher Tiemann (Phys., Univ. of Louisiana at Lafayette, Austin, Texas)

Passive acoustic monitoring (PAM) makes it possible to obtain reliable observations of marine mammals and to estimate the population density based on detected acoustic cues. PAM surveys offer higher accuracy density estimates than traditional visual surveys as long as the survey design is adequate and the probability of detection is reliably measured. The probability of detection depends on regional bathymetry, the season, the PAM system used, the detection algorithm, and the animal's acoustic apparatus. To improve the accuracy of the density estimation, this study focuses on understanding the relationship between the probability of detection of beaked whales and the detection algorithm used. The study utilizes the PAM data collected by fixed moored stations in the Gulf of Mexico in 2015 and 2017. The detectability function derived from experimental data is compared with the one obtained by modeling for two species of beaked whales (Cuvier and Gervais whales). The results will provide the guidance when and how modeling can be used to obtain reasonable estimates of the probability of the detection function. [Work supported by a grant from The Gulf of Mexico Research Initiative.]

9:30

4aAB5. Effects of click rate on bottlenose dolphin auditory brainstem response signal-to-noise ratio. James J. Finneran (SSC Pacific Code 71510, U.S. Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), and Robert F. Burkard (Univ. at Buffalo, Buffalo, NY)

Maximum Length Sequence (MLS) and Iterative Randomized Stimulation and Averaging (I-RSA) methods allow auditory brainstem response (ABR) measurements at high stimulus rates; however, it is not clear if high rates allow ABRs of a given signal-to-noise ratio (SNR) to be measured in less time than conventional averaging at lower rates. In the present study, ABR SNR was examined in six bottlenose dolphins using conventional averaging at rates of 25 and 100 Hz and the MLS/I-RSA approaches from 100 to 1250 Hz. Residual noise in the averaged ABR was estimated using root-mean-square values of the: waveform amplitude following the ABR, waveform amplitude after subtracting two subaverage ABRs, and amplitude variance at a single time point. For all approaches, residual noise decreased with the increasing measurement time. For a fixed recording time, SNR was highest at rates near 500 Hz, but optimal SNRs were only a few dB higher than that for conventional averaging at 100 Hz. Nonetheless, small improvements in SNR could result in significant time savings in reaching criterion SNR. The time savings allowed by the MLS and I-RSA methods will be discussed for both mean and individual data. [Work supported by U.S. Navy Living Marine Resources Program.]

9:45

4aAB6. Human auditory discrimination of bottlenose dolphin signature whistles masked by noise: Investigating perceptual strategies for anthropogenic noise pollution. Evan L. Morrison and Caroline M. DeLong (Dept. of Psych., Rochester Inst. of Technol., 18 Lomb Memorial Dr., Eastman 2309, Rochester, NY 14620, e.morrison@mail.rit.edu)

Anthropogenic masking noise in the world's oceans is known to impede many species ability to perceive acoustic signals, but little research has addressed how this noise pollution affects the detection of bioacoustic signals used for communication. Bottlenose dolphins use signature whistles which contain identification information. Past studies have shown that human participants can be used as models for dolphin hearing, but most previous research investigated echolocation. In experiment 1, human participants were tested on their ability to discriminate among signature whistles from three dolphins. Participants' performance was nearly errorless (M = 98.8%). In experiment 2, participants identified signature whistles masked by five different samples of boat noise, with different signals to noise ratios. Preliminary results suggest that participants perform worse in lower ratios of signal to noise, that some signature whistles are easier to identify in the presence of noise, and that some noises have more detrimental impacts on whistle recognition. The presence of boat noise may cause participants to use more auditory cues in order to identify whistles, although participants always relied most heavily on frequency contour and duration. This study may provide insight into the impacts of different types of boat noise on dolphin whistle perception.

Session 4aBAa

Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Inverse Problems in Biomedical Ultrasound I

T. Douglas Mast, Cochair
Biomedical Engineering, University of Cincinnati, 3938 Cardiovascular Research Center, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Kang Kim, Cochair
Medicine, University of Pittsburgh, 950 Scaife Hall, 3550 Terrace Street, Pittsburgh, PA 15261

Chair's Introduction—7:55

Invited Papers

8:00

4aBAa1. Acoustic holography for calibration of ultrasound sources and *in situ* fields in therapeutic ultrasound. Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Leninskie Gory, Moscow State Univ., Moscow 119991, Russia and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA, oa.sapozhnikov@physics.msu.ru), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Sergey A. Tsysar, Dmitry A. Nikolaev (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Leninskie Gory, Moscow State University, Moscow 119991, Russia and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA)

Therapeutic ultrasound sources, which are typically piezoelectric transducers, are intended to deliver known acoustic pressures to targeted tissue sites. Each transducer vibrates in a unique way and radiates a corresponding 3D ultrasound field. Accordingly, transducer vibrations should be known accurately in order to characterize the pressures delivered to the patient. Acoustic holography is a technique that relies on hydrophone measurements to reconstruct a source hologram that characterizes transducer vibrations [Sapozhnikov *et al.*, *JASA*, **138**(3), 1515–1532 (2015)]. In this way, a hologram is a signature of each transducer that can be monitored over time for quality assurance. Using holography-defined source boundary conditions, numerical forward projection of the ultrasound field based on the nonlinear wave equation can be used to accurately predict *in situ* temperatures and pressures in heterogeneous media for treatment planning. As such, acoustic holography goes beyond simple hydrophone scans and is uniquely suited to meet clinical needs for quantifying therapeutic ultrasound fields. In this paper, several examples of acoustic holography implementation are presented, including the characterization of single-element and multi-element flat and spherically curved sources working in linear and nonlinear regimes and in continuous and pulsed modes. [Work supported by NIH 1R01EB025187, R01EB007643, and R21CA219793; RFBR 17-02-00261 and 17-54-33034.]

8:20

4aBAa2. Full wave 3D inverse scattering: 21st century technology for whole body imaging. James Wiskin, Bilal Malik, Rajni Natesan, Nasser Pirshafiey, Mark Lenox, and John Klock (R&D, QT Ultrasound, LLC, 3 Hamilton Landing, Ste. 160, Novato, CA 94949, james.wiskin@qtultrasound.com)

Quantitative high resolution (QHR) images of speed of sound and attenuation in human breast have been made using full wave inverse scattering in three-dimension (FWIS3D), where only soft tissue is present. The FWIS3D technology and method are reviewed. Recent QHR images in the presence of bone and gas have been obtained with FWIS3D and are shown. Transmission mode quantitative and refraction corrected reflection images of small piglet abdomen, thorax, and head are shown. QHR images of the human knee using the same technology are shown. Human Knee is difficult due to the predominant presence of bone. With low frequency FWIS3D, the meniscus, structure within the Femur-Tibia (F-T) space, ligaments, and the infrapatellar fat pad can be seen. The intra-condyle space in the Femur is visible. It was earlier established that 3D modelling was necessary for breast. It is shown to be even more important for F-T space and whole body imaging. Quantitative estimates of high speed early development bone are made, and imaging through neo-natal skull is performed. Clear correspondence with known structures even in the presence of gas is displayed. This reveals FWIS3D ultrasound tomography as a 21st century whole body imaging modality.

8:40

4aBAa3. Iterative image reconstruction algorithm for transcranial photoacoustic tomography applications. Joemini Poudel (Dept. of Biomedical Eng., Washington Univ. in St. Louis, 6648 Oakland Ave., Saint Louis, MO 63139, jpoudel@wustl.edu), Lihong Wang (Andrew and Peggy Cherng Dept. of Medical Eng., California Inst. of Technol., Pasadena, CA), and Mark Anastasio (Dept. of Biomedical Eng., Washington Univ. in St. Louis, St. Louis, MO)

Photoacoustic computed tomography (PACT) is an emerging computed imaging modality that exploits optical contrast and ultrasonic detection principles to form images of the absorbed optical energy density. The PACT reconstruction problem corresponds to recovering the total absorbed optical density within a tissue sample, from the acoustic waves recorded on a measurement aperture located outside the support of the tissue sample. A major challenge in transcranial PACT brain imaging is to compensate for aberrations in the measured photoacoustic data due to their propagation through the skull. To properly account for these effects, a wave equation-based iterative reconstruction algorithm that can model the heterogeneous elastic properties of the medium is employed. To accurately recover the absorbed optical energy density, complete knowledge of the spatial distribution of the elastic parameters of the medium is required. However, estimating the elastic properties of the medium prior to the experiment is practically infeasible. To circumvent this, we propose to jointly reconstruct the absorbed optical energy density and the spatial distribution of the elastic parameters of the medium from PACT data alone. Reconstructed images from both numerical phantoms and experimental data are employed to demonstrate the feasibility and effectiveness of the approach.

9:00

4aBAa4. Direct error in constitutive equation formulation for inverse problems in time harmonic elastography. Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), Olalekan A. Babaniyi (Appl. Mathematics, UC Merced, Merced, CA), and Sanjay S. Yengul (Mech. Eng., Boston Univ., Brookline, MA)

Shear wave imaging techniques allow the evaluation of rigidity and viscosity of tissues locally within a material. From an inverse problem perspective, the approach is quite attractive insofar as it provides a densely sampled displacement field in the interior of the object from which to invert for material properties. We consider several challenges related to elastic wave inverse problems arising in acoustic radiation force imaging. First, we validate an axisymmetric viscoelasticity model suitable for some applications of acoustic radiation force imaging. Second, we consider reconstructing lateral displacement components from measured axial displacement components. Finally, we present a new variational formulation, the direct error in constitutive equation formulation, for inverse problems in time harmonic viscoelastic wave propagation with full-field data. The formulation relies on minimizing the error in the constitutive equation with a momentum equation constraint. Numerical results on model problems show that the formulation is capable of handling discontinuous and noisy strain fields and also converging with mesh refinement for continuous and discontinuous material property distributions. Applications to MRE and ARFI measured wave data are considered.

9:20

4aBAa5. Assessing FES-induced muscle fatigue using ultrasound to determine the inverse neuromuscular model for optimal FES input. Kang Kim (Medicine, Univ. of Pittsburgh, 950 Scaife Hall, 3550 Terrace St., Pittsburgh, PA 15261, kangkim@upmc.edu), Zhiyu Sheng, and Nitin Sharma (Mech. Eng. and Mater. Sci., Univ. of Pittsburgh, Pittsburgh, PA)

Functional electrical stimulation (FES) has been successful in activating paralyzed or paretic muscles to restore limb functions of individuals with impaired gait function. However, when activating the limb joint motion through externally stimulating muscle, rapid onset of muscle fatigue becomes a critical issue that results in injury. To overcome this challenge, an optimal FES input to the neuromuscular system needs to be determined and updated in real-time in order to maintain an effective, safe limb function. The inverse neuromuscular model between the desired limb joint motion and the FES input depends on time varying muscle contractility or fatigue level. In this study, ultrasound speckle tracking is proposed to assess muscle contractility and to establish a dynamic model. To demonstrate the feasibility, isometric knee extension experiments of healthy human participants were conducted with ultrasound imaging on the quadriceps muscle. The consistent decrease in peaks in strain and maximum knee joint torque during each contraction cycle suggest a potential correlation between the strain field and fatigue level of the target muscle. With further validation, ultrasound strain field can be used to solve for the dynamic neuromuscular model and further to determine the optimal FES input. Some technical challenges will also be discussed.

9:40

4aBAa6. Comparison of elastic modulus inverse estimation and the pulse wave velocity estimation for monitoring abdominal aortic aneurysms. Doran Mix, Luke Cybulski, Michael Stoner (Surgery, Univ. of Rochester Medical Ctr., Rochester, NY), and Michael S. Richards (Biomedical Eng., Rochester Inst. of Technol., 1 Lomb Memorial Dr., Rochester, NY 14623, msrbme@rit.edu)

The necessity of surgical intervention of abdominal aorta aneurysms is based on a risk-reduction paradigm primarily relying on trans-abdominal ultrasound (US) measurements of the maximum diameter of an AAA. However, the AAA diameter is only a rough estimate of rupture potential, and elastographic estimates of material property changes within aortic tissue may be a better predictor. This work compares an elastic imaging technique measuring aortic tissue stiffness in cross-section to a pulse wave velocity (PWV) estimate obtained from longitudinal images of the same geometry using a two-dimensional clinical US machine. The elastic imaging technique uses a linear elastic finite-element model to solve the elastic inverse problem and estimates the shear modulus. This technique uses a non-invasive pressure cuff to estimate the pressure in the aorta and normalizes the modulus values. The PWV technique uses geometric measurements and simplifies assumptions to create a direct relation between the wave speed and the modulus. Results of validation studies using aortic mimicking phantoms comparing modulus obtained from each of the techniques are presented. Initial clinical results will be also be presented.

10:00–10:15 Break

10:15

4aBAa7. Super-resolution ultrasound imaging for *in vivo* microvasculature assessment in acute kidney injury mouse model. Qiyang Chen (BioEng., Univ. of Pittsburgh, 3550 Terrace St., 624, Pittsburgh, PA 15261, chenqiyang0404@gmail.com), Brittney M. Rush (Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Jaesok Yu (BioEng., Univ. of Pittsburgh, Pittsburgh, PA), Roderick Tan, and Kang Kim (Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

Acute kidney injury (AKI) is a disease with a high mortality rate and increasing incidence. It also generates a high risk of developing into chronic kidney disease (CKD). The deterioration from AKI to CKD is associated with rarefaction of microvasculature in renal cortex. However, there is lack of well-established diagnostic method that can evaluate the microvasculature changes noninvasively and conveniently with a high spatial resolution during the progression from AKI to CKD. Ultrasound super-solution imaging is an emerging technology that can achieve a high spatial resolution of the vasculature beyond the acoustic diffraction limit by localizing the center of the signals from microbubbles. In this study, deconvolution based super-resolution ultrasound imaging is used to noninvasively assess the microvasculature changes in mouse kidney after AKI. Ultrasound scans on mouse kidneys at 3 weeks and 6 weeks post ischemia-reperfusion injury are compared with control mouse kidneys. Obvious microvasculature reduction due to AKI is identified, which is evidenced by histology. The feasibility of ultrasound super-resolution imaging as a potential diagnostic method for progressive renal disease after AKI is demonstrated in the AKI mouse model.

10:30

4aBAa8. Efficient sub-diffraction passive cavitation imaging. Scott J. Schoen, Zhigen Zhao (Mech. Eng., Georgia Inst. of Technol., 901 Atlantic Dr. NW, Rm. 4125K, Atlanta, GA 30318, scottschoenjr@gatech.edu), and Costas Arvanitis (Mech. Eng. and Biomedical Eng., Georgia Inst. of Technol. and Emory Univ., Atlanta, GA)

Acoustic localization of microbubbles offers a unique method to assess vascular structure and function noninvasively. To this end, passive imaging of the acoustic cavitation with the angular spectrum method (AS-PCI) is appealing as it is inherently fast and frequency-selective and thus allows stable cavitation activity to be isolated from other scatters via the bubbles' harmonic emissions. However, diffraction imposes a physical limitation on the resolution of acoustic imaging systems, which is typically on the order of millimeters for PCI. To enable rapid visualization of vessel structures with diameters of few hundreds of microns, we present a technique based on the AS method for fast super-localization (SL) of multiple, spatially separated bubbles that is 100-fold more efficient than time domain techniques employed for resolution improvement. We demonstrate, via experiments and numerical simulations, that it is possible to super-localize multiple bubbles within a single image and resolve vessels with diameters 10 times smaller than the diffraction limit (300 μm vs. 3 mm, respectively). Furthermore, successive super-localization of hundreds of microbubbles with the proposed SL-AS-PCI method allowed visualization of three-dimensional vessel structures within a few seconds on ordinary hardware. SL-AS-PCI holds great promise for efficient diagnosis of diseases associated with abnormal vasculature.

10:45

4aBAa9. Orientation-dependent anisotropy of acoustic properties of tendons at micrometer scale. Takuya Ogawa (Graduate School of Sci. and Eng., Chiba Univ., Chiba, Japan), Bin Yang, Po Lam (Dept. of Opthalmology, Univ. Of Pittsburgh, Pittsburgh, PA), Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan), Ian A. Sigal (Dept. of Opthalmology, Univ. Of Pittsburgh, Pittsburgh, PA), and Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org)

Tendons are bands of fibrous connective tissue connecting muscles to bones. They are composed of parallel arrays of collagen fibers closely packed together which makes them highly anisotropic. The anisotropy of the acoustic properties of tendons was investigated at an ultra-fine resolution (< 7 μm) using quantitative acoustic microscopy (QAM). Chicken tendons were fixed (formalin) while loaded longitudinally, then cryosectioned (16-μm) at several orientations (every 15 deg) from parallel (i.e., 0 deg) to perpendicular to the fibers (i.e., 90 deg). Two regions of two sections per angle were scanned using a QAM system operating at a center frequency of 250 MHz yielding a total of 28 QAM datasets which were processed to yield two-dimensional (2D) maps of the bulk modulus, mass density, acoustic impedance, and speed of sound for each scanned region. Acoustic parameters were averaged within each 2D map and mean and standard deviations computed at each angle. Results demonstrated a strong acoustical anisotropy. For instance acoustic impedance increased from 1.68 ± 0.08 to 1.87 ± 0.19 MRayl between 0 and 75 deg. Similarly, the speed of sound increased from 1686 ± 90 to 1958 ± 186 m/s between 0 and 75 deg. These results demonstrate the value of QAM to investigate the anisotropy of tissue microstructure and pave the way for using it to characterize other soft tissues with complex three-dimensional fiber orientations. [Work supported in part by NIH Grants EY023966 and EY028662.]

11:00

4aBAa10. Echo-mode aberration tomography: Sound speed imaging with a single linear array. Anthony Podkowa and Michael Oelze (Beckman Inst., 1009 W. Clark St. Apt. 205, Urbana, IL 61801, tpodkow2@illinois.edu)

Tomographic sound speed imaging has previously demonstrated the capability of producing images of comparable quality to that of X-ray CT and MRI. Traditionally, such reconstructions have only been achievable in transmission mode, either using diametrically opposed linear arrays or ring arrays. This is due to the conventional wisdom that forward scatter data are necessary for reconstruction in the general case, and consequentially, such setups are typically limited to easily externalized, soft tissues such as the female breast and thus are impractical for clinical usage. Recently, it has been demonstrated that in the presence of diffuse scatterers (Jaeger, 2015), pulse-echo reconstructions of slowness (inverse sound speed, proportional to refractive index) is feasible with a conventional single conventional linear array. By correlating data acquired with steered plane wave transmissions, depth dependent maps of phase lags can be generated and subsequently used to solve a multilinear inverse problem. The resulting images allow for baseband, speckle-free characterization of the underlying medium, which is complementary to the data acquired in traditional B-mode ultrasound. In this presentation, the fundamentals of echo-mode aberration tomography will be reviewed, completely with algorithmic formulation, beamformation considerations, and current challenges in practical reconstruction.

11:15

4aBAa11. Detection and imaging of prostate cancer using acoustic radiation force impulse imaging and quantitative ultrasound. Hong Chen (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), D. Cody Morris (Dept. of Biomedical Eng., Duke Univ., Durham, New York), Thomas Polascik (Dept. of Surgery, Duke Univ. Medical Ctr., Durham, NC), Wen-Chi Foo (Dept. of Pathol., Duke Univ. Medical Ctr., Durham, NC), Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York City, NY), Mark L. Palmeri, Kathryn Nightingale (Dept. of Biomedical Eng., Duke Univ., Durham, NC), and Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org)

Transrectal ultrasound (TRUS) imaging is routinely performed to guide core-needle biopsies for the definitive diagnosis of prostate cancer (PCa). Needles are rigidly attached to the transducer and randomly sample the prostate yielding a high rate of false negative determinations. This study investigated the combined use of acoustic radiation force impulse imaging (ARFI) and quantitative ultrasound (QUS) to detect cancerous lesions in the prostate during TRUS imaging. Three-dimensional (3D) RF data from 12 PCA patients scheduled for radical prostatectomy were acquired over the full gland volume during ARFI data acquisition and used to obtain effective

scatterer size (ESS), effective acoustic concentration (ESC), nakagami shape (μ), and nakagami scale (Ω) parameters. In each three-dimensional dataset, healthy and cancerous regions were obtained by manual segmentation using whole-mount histology slides. Linear discriminant and ROC methods were used to quantify the performance of ARFI displacements and QUS estimates at detecting PCa. Results for ARFI displacement and μ alone yielded an area under the ROC (AUC) curve of 0.84 and 0.69, respectively. The AUC value increased to 0.86 when μ and ARFI displacement were linearly combined. These results suggest that QUS and ARFI methods are sensitive to tissue properties affected by PCA. The proposed methods pave the way for novel real-time imaging of PCA during TRUS imaging. [Work supported in part by NIH Grants EB026233 and CA142824 and DOD PRCP Grant W81XWH-16-1-0653.]

11:30

4aBAa12. An Artificial Neural Network (ANN) approach to extract micro-architectural properties of cortical bone using ultrasound attenuation: A numerical study. Kaustav Mohanty, Omid Yousefian, Yasamin Karbalaeisadegh, Micah Ulrich, and Marie M. Muller (Dept. of Mech. and Aerosp. Eng., College of Eng., North Carolina State Univ., 3147 B, 911 Oval Dr., EB-3, Raleigh, NC 27606, kmohant@ncsu.edu)

The goal of this study is to estimate the porosity parameters including pore diameter, pore density, and porosity of cortical bone from ultrasound attenuation measurements using an artificial neural network (ANN). Two-dimensional (2D) finite-difference time-domain simulations are conducted to calculate the frequency-dependent attenuation in the range of 1–8 MHz in mono-disperse structures (constant pore size) with a pore diameter and density ranging from 20 to 120 μm and 3–16 pore/mm^2, respectively. Furthermore, poly-disperse structures (non-uniform pore distribution) are obtained from high resolution CT scans of human cortical bone and 2D numerical simulations are carried out in the same frequency range as for the mono-disperse cases. Then, a regression problem is formulated with the ultrasonic attenuation at different frequencies acting as the feature vectors and the output being set as the porosity parameters. Our dataset consists of 330 structures for the mono-disperse model and 668 structures for the poly-disperse model. ANN-based (3 hidden layers with 806 trainable weights) parameter prediction method achieves accuracies as high as 96% for pore size, 97% for porosity, and 78% for pore density for the poly-disperse model. This work demonstrates the potential of combining ultrasound methods to deep neural networks to quantify cortical bone parameters with high accuracies.

11:45

4aBAa13. Inferring elastic moduli of drops in acoustic fields. Jesse Batson, Rebekah Davis, and R. Glynn Holt (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jbats@bu.edu)

Acoustically levitated drops serve as non-contact mini-laboratories from which one can infer material properties from the response of the drop to the acoustic radiation force. Oddly enough, the oscillatory problem is more well-developed than the static problem. Analysis of the static acoustic deformation of Newtonian liquid drops is well established, yielding the inference of the surface tension. But the static deformation of an elastic drop is less well studied. The present work aims to enable the inference of elastic moduli from static deformations of acoustically levitated drops. The drop will be modeled as an incompressible, linear elastic solid undergoing small axisymmetric deformations. The axisymmetric interior stress and displacement fields will be found using Love's strain potential. The traction boundary condition can be calculated using linear acoustic theory . The measured static deformation of experimentally levitated drops with known material properties (polymer and protein gels) will be compared to the predictions of the theory. Time permitting, a finite element computational model will also be employed for comparison.

THURSDAY MORNING, 16 MAY 2019 BREATHITT, 8:00 A.M. TO 11:00 A.M.

Session 4aBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics I

Robert J. McGough, Cochair
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Hong Chen, Cochair
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Contributed Papers

8:00

4aBAb1. A parametric evaluation of shear wave speeds estimated with the time-of-flight approach in viscoelastic media. Luke M. Wiseman (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, wisemanl@msu.edu), Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN), and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

The time-of-flight approach estimates the shear elasticity in tissue mimicking elastography phantoms and in soft tissue. The time-of-flight approach is effective in elastic phantoms, but the time-of-flight approach tends to overestimate the shear elasticity in viscoelastic phantoms and in viscoelastic soft tissues. To characterize errors in estimated parameters for different values of the shear elasticity and the shear viscosity, three-dimensional (3D) shear wave simulations are evaluated for twelve different parameter combinations. The 3D acoustic radiation force is calculated for an L7-4 transducer using the fast nearfield method and the angular spectrum approach, and then, 3D shear wave propagation in a viscoelastic medium is simulated with Green's functions for a Kelvin-Voigt model. The time-of-flight method is then evaluated within a two-dimensional plane. The results show that the accuracy of the time-of-flight method depends on the values of the shear elasticity and the shear viscosity. In particular, the error in the estimated shear

elasticity increases as the shear viscosity increases, where the largest errors are observed when larger values of the shear viscosity are combined with smaller values of the shear elasticity. [Work supported in part by NIH Grants DK092255, EB023051, and EB012079.]

8:15

4aBAb2. Measured power law attenuation of shear waves in swine liver. Steven A. Grosz (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, groszste@msu.edu), Rebeca Pereira, Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN), and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

To characterize the attenuation and dispersion behavior of shear waves in excised swine liver samples, the complex shear modulus was measured with a Rheospectris C500+ from 10 Hz to 2000 Hz. The shear wave attenuation and shear wave speed were calculated from the complex modulus measurements. A power law fit was evaluated for the shear wave attenuation, and a power law fit with and without a constant offset was evaluated for the shear wave speed. The power law closely matches the measured shear wave attenuation over most of the frequency range evaluated, although some differences are observed in several of the samples below 400 Hz. The power law without the constant offset closely matches the measured shear wave speed above 200 Hz in all measurements, where the addition of the constant offset achieves a much closer fit in all measurements that contain discrepancies below 200 Hz. The results demonstrate that shear wave attenuation in swine liver follows a power law and that a power law with a constant offset is an effective model for the shear wave speed in swine liver. [Work supported in part by NIH Grants DK092255, EB023051, and EB012079.]

8:30

4aBAb3. Power law attenuation modeled as multiple relaxation. Sverre Holm (Informatics, Univ. of Oslo, Gaustadalleen 23B, Oslo N 0316, Norway, sverre@ifi.uio.no)

Wave equations with non-integer order derivatives may model power law behavior in medical and sediment acoustics. As experiments only support a finite bandwidth, there is a limit to how much physical insight that can be gained from such models. Other ways to model a power law are with a fractional heat law, hierarchical ladder models for polymer chains, and the non-Newtonian rheology of grain shearing. Multiple relaxation processes may be motivated by a hierarchy of substructures at different scales. It is also inherent in soft glassy materials, such as cells, with disordering and metastability. Even the Biot model with contact squirt flow and shear drag (BICSQS) may be interpreted as a multiple relaxation model. A weighted sum of relaxation processes will approximate a power law over a limited band, and an even distribution of relaxation frequencies on a logarithmic frequency axis, and with equal relaxation strengths, will give a power law attenuation with unit power, y = 1. This can be generalized to other power laws if the contribution from each relaxation process varies in proportion to the relaxation frequency to the power of y - 1. This scale-invariant distribution may hint at some fractal medium properties.

8:45

4aBAb4. Approximate analytical time-domain Green's functions for space-fractional wave equations. Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), James F. Kelly (Dept. of Statistics and Probability, Michigan State Univ., East Lansing, MI), and Madison Carriere (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, carrie51@msu.edu)

Approximate and exact time-domain Green's functions are available for time-fractional wave equations that describe power law attenuation in soft tissue, where each expression contains a stable probability distribution function. Previous work has also demonstrated that the exact time-domain Green's functions for time-fractional and space-fractional wave equations that describe power law attenuation are similar. Approximate analytical time-domain Green's functions have recently been derived for the Chen-Holm and Treeby-Cox space-fractional wave equations, where the approximate time-domain Green's function for the Chen-Holm wave equation contains a symmetric stable probability distribution function and the approximate time-domain Green's function for the Treeby-Cox wave equation contains a maximally skewed stable probability distribution function. Comparisons between the exact numerical and approximate analytical expressions for these time-domain Green's functions are evaluated for published values of the power law exponent and attenuation constant for breast and for liver. The results for both breast and liver converge very close to the source, and similar performance is observed in time-domain Green's functions computed for linear with frequency attenuation. Despite minor differences in the arguments, the approximate analytical time-domain Green's functions derived for dispersive time-fractional and space-fractional wave equations are also quite similar. [Work supported in part by NIH Grants EB023051 and EB012079.]

9:00

4aBAb5. Validity of Independent Scattering Approximation (ISA) to measure ultrasonic attenuation in porous structures with mono-disperse random pore distribution. Omid Yousefian, Yasamin Karbalaeisadegh, and Marie M. Muller (North Carolina State Univ., 2704 Brigadoon Dr., Apt A, Raleigh, NC 27606, ykarbal@ncsu.edu)

The goal of this study is to assess the validity of the Independent Scattering Approximation (ISA) for predicting ultrasonic attenuation in structures mimicking simplified geometries of cortical bone. Finite Difference Time Domain (FDTD) methods were used to assess the ultrasound attenuation in porous media with a monodisperse distribution of pores, with pore diameters, density, and frequency in the range of $\phi = 40$–120μm, 3–16 pore/mm^2, and 1–8 MHz, respectively. The attenuation values obtained from the FDTD simulations were compared to attenuation values predicted by the ISA. The results indicate that the ISA reliably predicts the attenuation for $k\phi<1$ and $\phi\in[100,120]$ μm, with less than 15% error. The error increases up to 26% as ϕ decreases. The reason that ISA fails to predict accurate values for lower ϕ is investigated through the quantification of multiple scattering (MS). This is done by MS assessment in which the effect of multiple versus single scattering (SS) is compared by measuring the backscattered signals on a simulated linear array transducer. <!– Give a bit more details on how you compared the MS and SS –>The results revealed that MS is dominant at $\phi = 120$, but that SS is dominant for $\phi = 60$ μm. Assuming that the attenuation is a function of $k\phi$, the ISA is modified to test its applicability where single scattering is dominant. The results using the modified ISA showed that it can predict the attenuation in monodisperse porous structures for $k\phi < 1$ and $\phi\in[40$–$100]$ μm with less than 10% error.

9:15

4aBAb6. *In situ* calibration to account for transmission losses in backscatter coefficient estimation. Trong Nguyen (Univ. of Illinois at Urbana-Champaign, 405 N Mathews Ave., Urbana, IL 61801, tnnguyn2@illinois.edu), Alex Tam (Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Michael L. Oelze (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The backscatter coefficient (BSC) has demonstrated the ability to classify disease state and identify the response of cancer to therapy. However, estimating the BSC *in vivo* using a reference phantom technique does not account for transmission losses due to intervening layers, leading to increase in bias and variance of BSC-based estimates from one sample to the next. To account for transmission losses, an *in situ* calibration approach is proposed using a titanium sphere that is well-characterized ultrasonically, biocompatible, and embedded inside the sample. Ultrasound scattered from the sphere encounters the same transmission loss and attenuation as the investigated sample and can be used as a reference spectrum. To test the calibration procedures, phantoms were scanned with and without lossy layers on top, and BSCs were estimated using the *in situ* calibration approach and the reference phantom approach and compared. The differences of the BSCs, using the BSC from the reference phantom without a layer as baseline, were 0.16 ± 2.29 dB, -1.95 ± 2.99 dB and -10.90 ± 3.64 dB using the *in situ* calibration approach without the layer, with the layer, and using the reference phantom approach with the layer, respectively. The results indicate that an *in situ* calibration target can account for overlaying tissue losses thereby improving the robustness of BSC-based estimates.

9:30–9:45 Break

9:45

4aBAb7. Investigation into tendon histotripsy. Molly Smallcomb and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, molly.smallcomb@gmail.com)

Histotripsy uses cavitation bubble clouds or shock wave heating and millisecond boiling to fractionate soft tissues. While this modality has proven successful in debulking most soft tissues, highly collagenous tissues such as tendons have proven resistant to mechanical fractionation using histotripsy. In this study, *ex vivo* rat and bovine Achilles tendons were placed at the focus of a 1.5-MHz transducer and exposed to 1–20 ms pulses repeated at 1 Hz for 1 min over ranges of acoustic pressures up to $p+ = 88$ MPa (peak positive), $p = 20$ MPa (peak negative). Simultaneous ultrasound imaging with the Verasonics® research ultrasound system and ATL L7-4 transducer monitored bubble activity, or hyperechogenicity, during the histotripsy exposure. Collected samples were stained with Hematoxylin and Eosin for histological analysis of tissue disruption. Preliminary results show hyperechogenicity within the tendon during the histotripsy exposure; however, thus far only thermal injury has been found histological. The threshold to detect hyperechogenicity in the tendon for 10-ms pulses were $p+ = 63$ MPa, $p = 19$ MPa. Future work involves additional parameter testing to promote mechanical fractionation rather than thermal injury of tendons. [Work supported by Penn State College of Engineering Multidisciplinary Research Seed Grant]

10:00

4aBAb8. Toroidal intra-operative high intensity focused ultrasound transducer for treating liver metastases under ultrasound imaging guidance: Clinical results of Phase II study. David Melodelima (LabTAU - INSERM U1032, 151 cours Albert Thomas, Lyon 69003, France, David.Melodelima@inserm.fr), Aurelien Dupre, Yao Chen, David Perol, and Michel Rivoire (Ctr. Leon Berard, Lyon, France)

The aim of this study was to assess the feasibility, safety, and accuracy of HIFU ablation in patients with liver metastases in a prospective phase II trial. The transducer has a toroidal shape (diameter: 70 mm, radius of curvature: 70 mm) and was divided into 32 ring-shaped emitters operating at 2.5 MHz. Thirty-one patients were included. HIFU ablations were created to ablate metastases (up to 30 mm in diameter) with safety margins in all directions. The use of a toroidal transducer enables an ablation rate (10 cc min^{-1}) significantly higher than spherical transducers. Therefore, using electronic focusing of the beam, it was possible to treat all metastases with safety margin without the need to displace the device between HIFU exposures. The exposure time varied from 40 s to 370 s according to the diameter of the metastases to be treated. The dimensions of these HIFU ablations were a diameter of 48 $\pm$ 4.9 mm and a long axis of 45 $\pm$ 3.4 mm. No damage occurred to neighboring tissues. This study is the first clinical use of intra-operative HIFU in patients with liver metastases.

10:15

4aBAb9. Design of a histotripsy array for the treatment of intracerebral hemorrhage. Tyler Gerhardson, Jonathan R. Sukovich, Jonathan E. Lundt, Ning Lu (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48105, tgerhard@umich.edu), Aditya Pandey (Dept. of Neurosurgery, Univ. of Michigan, Ann Arbor, MI), Charles A. Cain, Zhen Xu, and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Histotripsy is a focused ultrasound technique using short, high amplitude pulses to generate targeted cavitation. Recently, the feasibility to treat blood clots through human skulls with histotripsy has been shown *in vitro*. The purpose of this study was to evaluate acoustic parameters for developing an optimized array design for the treatment of intracerebral hemorrhage (ICH) with histotripsy. The main performance criteria were to achieve a large electronic focal steering range ($\geq$ 20 mm) and an ability to correct aberration through the skull. A hemispherical aperture of 150 mm radius was considered with modular elements to allow for arbitrary insertion of a catheter hydrophone to perform aberration correction. The attenuation at discrete frequencies from 250 kHz–2 MHz was measured through excised human skulls ($n = 7$) along with effects of incidence angle and aberration to optimize the operating frequency. Different piezoelectric materials were tested to optimize the peak-output, transduction efficiency, and durability at high PRF. One of the best performers, 17 mm square PZ36 material at 700 kHz, was found to produce at least 1.5 MPa free field at PRFs of 1 kHz. Finally, simulation showed that an optimally packed array configuration using 360 modules should be able to achieve an effective steering range of at least $\pm$20 mm through the skull.

10:30

4aBAb10. Focused ultrasound-mediated microbubble destruction for glioblastoma treatment. Lifei Zhu, Arash Nazeri, Michael Altman, Dinesh Thotala, Nima Sharifai (Washington Univ. in St. Louis, St. Louis, MO), and Hong Chen (Washington Univ. in St. Louis, 4511 Forest Park, St. Louis, MO 63108, hongchen@wustl.edu)

Glioblastoma is the most common primary brain tumor with a poor prognosis despite advances in various treatment modalities, such as radiation therapy (RT). This study compared the tumor growth inhibition effects of focused ultrasound-targeted microbubble destruction (UTMD) therapy with the RT using an orthotopic mouse glioma model. Mice were implanted with GL261 glioblastoma cells and divided into three groups: control group (no treatment); RT group (2 Gy/day, 5 days/week, 3 consecutive weeks); and UTMD group (FUS sonication in the presence of systemically injected microbubbles at the peak negative pressure of 1.5 MPa, frequency of 1.44 MHz, and 2 treatments/week for 3 consecutive weeks). Contrast-enhanced magnetic resonance imaging (MRI) was performed once every four days for measuring the tumor volume. Both UTMD and RT caused significant growth inhibition compared to the control group; however, there was no significant difference between the UTMD and RT groups. Terminal deoxynucleotidyl transferase dUTP nick end labeling (TUNEL) staining showed that the number of apoptotic tumor cells in both RT and UTMD groups were significantly higher than the control group without the difference between these two groups. This study suggests that UTMD suppressed glioblastoma tumor growth and this effect was comparable with that achieved by RT.

10:45

4aBAb11. *Ex vivo* thermal ablation monitoring using three-dimensional ultrasound echo decorrelation imaging. Elmira Ghahrahmani Z. (Biomedical Eng., Univ. of Cincinnati, 3960 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, ghahraea@mail.uc.edu), Peter D. Grimm (Elec. and Comput. Eng., Univ. of Cincinnati, Cincinnati, OH), Michael T. Cox, Kathryn J. Eary, E. G. Sunethra Dayavansha, and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Echo decorrelation imaging is a method for quantitatively mapping transient heat-induced changes in pulse-echo ultrasound images. For clinical thermal ablation of liver cancer using radiofrequency or microwave ablation (RFA or MWA), real-time three-dimensional (3D) echo decorrelation imaging is necessary because the entire tumor, with typical diameter 2–5 cm, is ablated at once. We present a method for constructing 3D echo decorrelation maps during radiofrequency ablation (RFA) in *ex vivo* bovine liver using beamformed in-phase and quadrature (IQ) echo acquired from a Siemens Acuson SC2000 scanner and 4Z1c matrix array. To directly compare echo decorrelation images to the desired outcome of tissue ablation, 3D echo decorrelation images are compared to volumetric reconstructions of the thermal ablation zone, obtained from optical scans of regularly spaced tissue sections. Capability of echo decorrelation as a predictor of local ablation is assessed using receiver operator characteristic curve analysis. Similar to previous studies of two-dimensional echo decorrelation imaging, good correspondence is seen between 3D echo decorrelation images and ablated tissue histology.

Session 4aEA

Engineering Acoustics: General Topics in Engineering Acoustics: Sensors and Sources

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Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758

Contributed Papers

8:00

4aEA1. Developments in underwater thermophones. Nathanael K. Mayo, Christian Schumacher, John B. Blottman, and Thomas R. Howarth (Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, RI 02841-1708, nathanael.mayo@navy.mil)

Thermophones are acoustic projectors powered by resistive heating elements. The thermophone was invented shortly after Alexander Graham Bell's invention of the telephone, and since then, thermophone development has primarily focused around its use in air. Thermophones have historically been used as a precision source of sound for calibrating microphones and have been unable to break out of this limiting role largely due to device inefficiencies. The recent implementation of nanomaterials, which greatly improve the overall electric to acoustic conversion efficiencies, has brought about a resurgence of interest surrounding these devices; however, practical applications remain elusive. One of the most likely fits for this unique technology is as an underwater projector. Recent calibrated acoustic testing on various designs of encapsulated underwater thermophones reinforces our notional assumptions of the electro-thermo-acoustic transduction process as well as the general effect of various device parameters on acoustic performance. While efficiency is still a major concern, thermophones possess many other desirable features such as their low cost, wide bandwidth, and ability to produce low frequencies in a compact package.

8:15

4aEA2. Focal power efficiencies achieved with aluminum Fresnel and fraxicon low-profile lenses. Robert L. Lirette (Phys., Univ. of Mississippi, 2400 Anderson Rd., Apt. 4, Oxford, MS 38655, rlirette@go.olemiss.edu) and Joel Mobley (Phys., Univ. of MS, University, MS)

Fresnel and fraxicon phase plate lenses modify incident acoustic radiation by a series of phasing steps. A Fresnel lens is designed to approximate a spherical lens, and a fraxicon is designed to approximate an axicon generating a non-diffracting Bessel beam out to a depth of focus. For this study, two lenses were manufactured out of aluminum, a Fresnel with a focal point of 5 cm and a fraxicon with a DOF of 10 cm. Each lens was bonded to a 1.2-MHz piezoelectric driver with epoxy and the assemblies mounted in PVC housings for use under water. The field from each was measured by a hydrophone scanned in the transverse planes. Using an acoustic force balance, the electromechanical conversion efficiency was measured to be roughly 50% for each. The focal power fraction, defined as the ratio of the power within the -6 dB boundary of the focus to the total emitted power, was determined for both transducers. The results are compared to numerical simulations and previously published results.

8:30

4aEA3. Computational viscothermal acoustic study of micro-electro-mechanical systems (MEMS) perforated plates. Vahid Naderyan, Richard Raspet, Craig Hickey, and Mohammad Mohammadi (National Ctr. for Physical Acoust., Univ. of Mississippi, NCPA, 1 Coliseum Dr., University, MS 38677, vnaderya@go.olemiss.edu)

Micro-perforated plates (MPP) are widely used as sound absorption materials in many noise control applications. Acoustic properties of the MPPs have been theoretically and experimentally studied for many years. The results of these studies are often used in the studies of MEMS devices with perforated plates. However, there exist differences in the physical dimensions of MPPs and MEMS perforated plates. The typical MPP perforation radius is in the range of 1 mm to 1 cm. For these dimensions and audio frequencies, the shear wave-number is much larger than 1. The dimensionless shear wave-number, which is an unsteady Reynolds number, is a measure for the ratio between inertial and viscous effects. Hence for typical MMPs, the inertial effects are dominant. However, the typical hole radius in the MEMS perforated plates is below 20 μm corresponding to sub-unit shear wave-numbers. Therefore, in MEMS perforated plates, the viscous effects are the dominant part of the impedance. In addition, typical MPPs have low porosities on the order of 1%, whereas typical MEMS perforated plates have high porosities in the range of 25% to 75%. In this work, viscous and thermal losses and also the end effects of the MEMS perforated plates are studied using the finite element method.

8:45

4aEA4. Miniature fiber optic acoustic pressure sensors with air-backed graphene diaphragms. Qian Dong and Haijun Liu (Mech. Eng., Temple Univ., 1947 N. 12th St., Philadelphia, PA 191226018, haijun@temple.edu)

Graphene has been known to possess exceptional mechanical properties, including its extremely high Young's modulus and atomic-layer thickness. Although there are several reported fiber optic pressure sensors using a graphene film, a key question that is not well understood is how the suspended graphene film interacts with the backing air cavity and affects sensor performance. Based on our previously analytical model, we will show that sensor performance suffers due to the significantly reduced mechanical sensitivity by the backing cavity. To remedy this limitation, we will, through experimental and numerical methods, investigate two approaches to enhance the sensitivity of fiber optic acoustic pressure sensors using the graphene film. First, a graphene-silver composite diaphragm is used to enhance the optical sensitivity by increasing the reflectivity. Compared with a sensor

with pure graphene diaphragm, a graphene-silver composite can enhance the sensitivity by three-fold, while the mechanical sensitivity is largely unchanged. Second, a fiber optic sensor is developed with enlarged backing air volume through the gap between an optical fiber and a silica capillary tube. Experimental results show that the mechanical sensitivity is increased by 10xfrom the case where the gap side space is filled.

9:00

4aEA5. Detection of remotely piloted aircraft using bio-acoustic techniques. Jian Fang, Michael Driscoll, Russell Brinkworth, and Anthony Finn (Defence & Systems Inst., Univ. of South Australia, Bldg. W, Mawson Lakes, SA 5095, Australia, anthony.finn@unisa.edu.au)

This paper describes a biologically inspired approach for acoustically detecting and tracking small remotely piloted aircraft based on processing found in the insect visual system. Previous work has shown the insect visual system is excellent at enhancing and isolating signals in complex and noisy visual scenes. By constructing spectrograms of audio signals, we essentially converted audio data into images, which could then be processed in the same way as visual data sets. Traditional time-frequency analysis was used to characterise the signatures of remotely piloted aircraft observed by multiple sets of small microphone arrays located on the ground. A model based on multiple layers of non-linear dynamic adaptive components measured from responses of insect visual neurons was then applied to the observed spectrograms to enhance the related acoustic harmonics and suppress the unrelated noise. The result was crisp low-amplitude signal detection and classification of these difficult target sets. In contrast to traditional systems that operate uniformly across the entire spectrum—attempting to capture the world as faithfully as possible—the bio-inspired processing uses multiple time scales and operates independently on each time-frequency cell ("pixel"). This reveals unseen harmonics otherwise hidden by noise, thereby extending the maximum range at which even slow-moving, low amplitude targets are detected and tracked.

9:15

4aEA6. A bio-inspired sound source localization sensor with internal coupling. Qian Dong and Haijun Liu (Mech. Eng., Temple Univ., 1947 N. 12th St., Philadelphia, PA 191226018, haijun@temple.edu)

The mechanism of using internal coupling to enhance directional hearing has been found in various animals across multiple length scales, including crickets, lizards, frogs, birds, and alligators. For each eardrum, the acoustic stimuli impinge not only on the front side but also on the opposing side via the connecting cavity. The combination of these two stimuli renders a much higher directional sensitivity than the case with two uncoupled independent receivers. Inspired by this mechanism found in Nature, here we present a bio-inspired sound source localization sensor which consists of two pre-tensioned membranes on a three-dimensional printed housing. The vibration of the two membranes is detected by a low coherence fiber optic interferometric system. The experimental results from this prototype will be demonstrated to validate the feasibility of developing miniature bio-inspired devices for sound source localization.

9:30

4aEA7. Multi-channel broadband receive array for downward looking sonar applications. Bryan D. Todd, Jermaine L. Kennedy, and David E. Malphurs (Naval Surface Warfare Ctr. Div., 110 Vernon Ave., Panama City Beach, FL 32407, Bryan.D.Todd@navy.mil)

The recent proliferation in interest for increased detection and classification probabilities of submerged objects in maritime environments has established a need for broadband underwater acoustic receivers. A multi-channel sonar receive array composed of 16 piezoelectric ceramic elements was designed, fabricated, and developed using rapid prototyping techniques including a combination of three-dimensional printed materials, molding, and casting techniques to support various modalities of underwater sonar sensing applications. Four sets of the receive elements, consisting of four individual elements per set, were mounted using disparate methods. In-water acoustical property characterizations over a broad operating frequency range were examined and analyzed. The aforementioned results fostered identification of key array design characteristics for super-critical grazing angle downward looking sonar (DLS) systems.

9:45–10:00 Break

10:00

4aEA8. Design of a conformal acoustic parametric array. Matthew Malone and Eric A. Dieckman (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, mmalo2@unh.newhaven.edu)

The acoustic parametric array exploits the nonlinearity of air to create an audible sound beam that can propagate long distances. Transmitted signals are modulated around a nominal 40 kHz carrier, creating sum and difference components as the signal propagates through air. Since attenuation is proportional to frequency squared only the low-frequency difference component remains at long distances. Current commercially available parametric arrays arrange ultrasonic transducers in a planar array to create a beam of sound that is audible at distances up to 100 m. Our goal is to create a concave parametric array to determine if the added geometrical focusing allows for tighter spatial control of the audible signal. Ultrasonic transducers were mounted on a flexible three-dimensional printed structure to create an array with a variable curvature. Simulation and experimental results are presented comparing our conformal array to two commercially available planar arrays.

10:15

4aEA9. Analysis of a passive radio frequency excited acoustic transducer. Charles Thompson (ECE, UMASS, 1 University Ave., Lowell, MA 01854, charles_thompson@uml.edu), Johnetta Jallah (Lowell HS, Lowell, MA), Grace Remillard, and Kavitha Chandra (ECE, UMASS, Lowell, MA)

In this paper, the acoustic sensitivity of passive transducers excited at radio frequency is examined. This wireless battery-free sensing platform derives its power from an externally applied electromagnetic field generated by a radio transmitter. The audio signal is encoded in the backscattered electromagnetic field. Electro-Mechano-Acoustical analogies are developed and presented. Power generation, sound transduction, and radio frequency backscatter transmission of the audio signal are examined.

10:30

4aEA10. Acoustic radiation characteristics improvement according to the shape change of flat-plate display exciter speaker. Hyung Woo Park (IT, Soongsil Univ., Seoul, South Korea), SungTae Lee, and Kwanho Park (LG Display, 245 LG-ro, Paju-si, Paju, South Korea, owenlee@lgdisplay.com)

For human, sound and video are evolving as the information transmission method. People are quick and easy to understand when sound and video are transmitted at the same time. In previous studies, we introduced a study to increase the quality of sound by equalizing the position of sound and sound in a flat panel display such as OLED (organic light emitting diode) TV. In that, we have implemented a sound field by separating two or more channels by vibrating the display panel of an OLED display. In this study, we investigated the effect of the variation of the shape of the pole piece on the radiation characteristics of the exciter speaker. In dynamic speakers, a pole piece guides the voice coil. However, in the exciter, the Pole Pieces excite the diaphragm directly. Initially, due to the limitations of exciter manufacturing technology, two exciters were placed separately. The first improvement was arranged on the same axis, and the sound was implemented by twin structure. However, because of the use of two pole pieces, the subtle phase difference of the two influenced the radiation pattern. In this study, the elliptical pole piece was introduced to improve the radiation characteristics in the transverse axis direction. The use of a single pole piece not only improves acoustical characteristics in the longer radius direction; furthermore, it is confirmed that the sound quality is improved by reducing manufacturing and driving errors even in the radial direction.

10:45

4aEA11. On the downward direction of the flat panel display speaker. Hyung Woo Park (Commun. Eng., Soongsil Univ., 520 Computing Bldg. 369 Snagdo-Ro, Dongjak-Gu, Seoul 06978, South Korea, pphw@ssu.ac.kr), SungTae Lee, Kwanho Park (Commun. Eng., Soongsil Univ., Paju, South Korea), and Myungjin Bae (Commun. Engeering, Soongsil Univ., Seoul, South Korea)

Information display devices are being advancement by ICT development. Particularly, besides the image quality and appearance design of the information display device, the development of the accompanying elements such as the sound quality is also progressed. Conventional information display devices such as LCD and LED TV have focused on the pixel configuration and color implementation and developed to such an extent that the human eye can not follow it, and the image quality is as good as the actual view in a proper field of view. However, in the case of sound, it has occurred at various alternative positions with the limitation that it cannot penetrate the hard screen. In the case of a flat-screen TV, however, the position of the sound was properly configured by reproducing the sound directly from the left and right sides of the screen. However, focusing on the image quality and design elements, it was hidden above. With several experimental factors, we were able to reproduce a lot of sound from the bottom speaker. This is disadvantageous in that it cannot hear the reproduced sound directly, hears mainly the reflected waves of the space below the space where the information display device is located, and hears different sounds depending on the characteristics of the reflection surface. In this study, we introduce a technique to make a sound with a direct screen that complements these shortcomings.

THURSDAY MORNING, 16 MAY 2019 SEGELL, 8:50 A.M. TO 12:00 NOON

Session 4aNS

Noise and Education in Acoustics: Increasing Noise Awareness in Society

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

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—8:50

Invited Papers

8:55

4aNS1. The international Noise Awareness Day in Germany. Brigitte Schulte-Fortkamp (TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

The International Noise Awareness Day in Germany was organized first time in 1998 and will have on 24 April 2019 his 22nd yearly event. The sensitization in relation to the problem of noise along with the spreading of knowledge about causes and consequences of noise (both socially and health wise) is elementary constituents of the "Tag gegen Lärm" (Noise Awareness Day). Through its continuity over the past 22 years and its public acceptance, the "Tag gegen Lärm" has become an institution that has a permanent place in Germany's calendar. The "Noise Awareness Day" is aiming everyone who is interested in noise, its causes, consequences, and countering, including people affected by noise, subject-specific interested groups, and people with political responsibilities (citizens, economy, and politics). The "Noise Awareness Day" happens every year in April, always scheduled coordinated with the "International Noise Awareness Day" organized by the Center for Hearing and Communication (CHC) USA. Current work and activities will be presented.

9:15

4aNS2. International Noise Awareness Day and the Acoustical Society of America: Past, present, and future. L. Keeta Jones (Acoust. Society of America, 1305 Walt Whitman Rd., Ste. 300, Melville, NY 11787, kjones@acousticalsociety.org)

Prior to 2018, the Acoustical Society of America (ASA) did little to support the Center for Hearing and Communication's International Noise Awareness Day (INAD). Task Force 1 (TF1) members of the five-year ASA Strategic Plan determined that ASA must make a greater effort to support the INAD campaign to help raise public awareness of noise to meet our own strategic goals. For 2018, TF1 members organized and promoted activities that would not only increase noise awareness but would also encourage the public to interact with the ASA. These activities included taking sound level measurements using a mobile app, reading a Proceedings of the Meetings on Acoustics paper, watching a movie, and taking part in a live YouTube discussion with expert panelists. The success of these activities is measured in increased downloads, website traffic, followers, and subscribers. To continue ASA's involvement with future INADs once the strategic plan ends, organizing was moved to the Technical Committee on Noise. This presentation will end with a summary of ASA INAD 2019, set to take place on Wednesday, 24 April 2019, as well as an update on future INAD plans.

9:35

4aNS3. Soundprint and the ASA's International Noise Awareness Day Campaign—Results, what worked, and going forward. Gregory S. Farber (SoundPrint, P.O. Box 74, New York City, NY 10150, greg@soundprint.co)

SoundPrint, a crowdsourcing app that objectively measures noise levels of venues and described as "Yelp for Noise," partnered with the Acoustical Society of America (ASA) for International Noise Awareness Day in 2018 and 2019 to raise noise pollution and hearing health awareness. SoundPrint served as the technological tool by which the public used to "via their sound level submissions" to show the public that noise is an important public health issue. The 2018 INAD campaign's initial success led to a significant bump in the number of crowdsourced submissions to SoundPrint's database spanning numerous countries, states, and cities (figures will be presented). The campaign was supported with specific marketing content on noise pollution and hearing health that raised visibility for both INAD and ASA. The presentation will discuss data results associated with the campaign, marketing methods used to raise awareness, and subsequent steps other organizations can employ to further ASA's cause going forward.

9:55

4aNS4. Centers for Disease Control and prevention efforts to increase awareness and prevention of noise-induced hearing loss. Yulia Carroll and John Eichwald (National Ctr. for Environ. Health, Ctr. for Disease Control and Prevention (CDC), 4770 Buford Hwy., CDC Chamblee - Bldg. 102 Rm. 2128, Atlanta, GA 30341, YCarroll@cdc.gov)

For 45 years, the Centers for Disease Control and Prevention (CDC) has researched noise induced hearing loss (NIHL) in the workplace and disseminated its research to prevent occupational hearing loss. Additionally, CDC has made research and educational materials available on hearing loss in children. In 2015, CDC received inquiries from the public and medical community about NIHL in non-workplace settings. In response, CDC began efforts to raise public awareness of NIHL and awareness about how to prevent its health effects. A CDC intra-agency working group collaborated with the World Health Organization, the National Institute on Deafness and Other Communication Disorders and the Dangerous Decibels® program for the promotion of the materials including (1) MMWR Vital Signs: Noise-Induced Hearing Loss Among Adults—United States 2011–2012; (2) CDC Public Health Grand Rounds: Promoting Hearing Health Across the Lifespan; (3) World Hearing Day educational materials; and (4) MMWR: Use of Personal Hearing Protection Devices at Loud Athletic or Entertainment Events Among Adults — United States, 2018. Additionally, CDC scientists and communicators continue to leverage internal and external channels for developing materials and spreading the word about the prevention of NIHL at work, at home and in communities.

10:15–10:30 Break

10:30

4aNS5. Total hearing health: An approach for raising noise awareness in society. Christa L. Themann (Hearing Loss Prevention Team, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., MS C-27, Cincinnati, OH 45226, clt6@cdc.gov)

Noise-induced hearing loss is one of the most common work-related illnesses in the United States, and an estimated 24% of hearing difficulty among the working population is attributable to workplace exposures. However, the harmful effects of noise are evident even among non-workers. Regardless of occupation, nearly everyone will encounter hazardous noise at some point during their lifetime. NIOSH promotes Total Hearing Health, which broadens the scope of hearing loss prevention interventions to encompass all risks to hearing, both at and away from work. This presentation will discuss tools for applying the Total Hearing Health approach to increasing noise awareness. These tools include (a) Apps and devices which measure noise exposure and provide information on hearing loss risk; (b) Promotional ideas to raise awareness of hearing health at worksites, classrooms, health fairs, sporting events, and other venues; and (c) Wikipedia, blogs, and social media tools for expanding the reach of hearing loss prevention messages. Increasing noise awareness requires engaging the public in an authentic and meaningful way. Examples of how NIOSH have implemented Total Hearing Health to accomplish this goal and recommendations for incorporating Total Hearing Health in your own work will be provided.

10:50

4aNS6. National occupational research agenda for hearing loss prevention. William J. Murphy (Hearing Loss Prevention Team, Ctr. 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) and Amanda S. Azman (Pittsburgh Mining Res. Div., Ctr. for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Pittsburgh, PA)

The National Institute for Occupational Safety and Health (NIOSH) has a mandate to conduct research on occupational safety and health. The research portfolio is organized by industrial sectors and cross-sectors for illnesses and injuries that are found in all sectors. The Hearing Loss Prevention research cross sector council comprises representatives from government, labor organizations, academia, and industry representatives. The HLP council held several meetings throughout 2018 to determine research needs for occupational hearing loss prevention in the United States. Five topic areas were determined. (1) Provide input for policies and guidelines that will inform best practices for hearing loss-prevention efforts. (2) Develop effective, evidence-based education designed to improve hearing conservation program outcomes for exposed workers and management. (3) Develop, commercialize, and widely implement noise control solutions on jobsites in key industries. (4) Develop audiological tests for hearing loss prevention. (5) Improve occupational hearing loss surveillance. These topic areas will be discussed in detail to help motivate other researchers to join further our knowledge to prevent occupational hearing loss.

Contributed Papers

11:10

4aNS7. Variation of sound events that stand out in one's memory across the ages: Comparison between 2008 and 2016. Takeshi Akita and Lei Qu (Dept. of Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Senju-Asahi-cho, Adachi-ku, Tokyo 1208551, Japan, akita@cck.dendai.ac.jp)

To contribute to create good soundscape, sound events that stand out in one's memory are surveyed. In the present research, the data of written questionnaire that is carried out in the coursework of acoustics is analyzed. Especially, results of 2016 are compared with that of 2008. In the questionnaire, students are instructed to write down sound events that are easily recalled at the moment, after they remember the sound events that they heard in the period from the time of awakening to the coursework. Additionally, students evaluate each sound event whether they have good or bad impression. As the result, average of recalled sound events was 10.1 per person in 2016 while it was 8.2 in 2008. Sound events are classified into three large categories at each year, and the composition of the classification is the same between 2016 and 2008. They are labeled as Sound produced by a person or people, Artificially produced sound, and Nature sound. In 2016 data, the number of artificially produced sound that has no good nor bad evaluation increases significantly in comparison to 2008. It is suggested that popularization of information technology and smartphone produces more electronic sound in the urban soundscape.

11:25

4aNS8. Public health impacts from subway noise: Case study Hong Kong. Stephany Y. Xu (Harvard Univ., Extension School, 51 Brattle St., Cambridge, MA 02138, syx440@g.harvard.com), Changyong Jiang, and Lixi Huang (Lab for AeroDynam. and Acoust., Dept. of Mech. Eng. and Zhejiang Inst. of Res. and Innovation, The Univ. of Hong Kong, Hong Kong, Hong Kong)

In cities, subway noise is often cited as a major contributor to noise pollution that impacts millions of people every day. Previous studies on this topic have shown that peak subway noise levels in some cities can be as high as 110 dB, which greatly exceeds the 70 dB level set by the World Health Organization (WHO) and EPA for safe environmental noise levels. This work aims to characterize the subway noise in Hong Kong, analyze potential source features, and make technical recommendations for consideration by government and metro companies. First, the overall noise data on all nine subway lines in the city are presented and compared with published data of other subway lines around the world. Spectra of the loudest segments are analyzed to show the effects of tunnel modes, track curvature, and other features that may play a significant role in noise radiation and reverberation. A detailed correlation study is conducted for the short-time noise level and vehicle speed. A new train speed profile that optimizes noise exposure reduction is proposed for consideration of a future autodrive system. Finally, a study of vibroacoustic exposure by passengers is also conducted to examine the impacts beyond the audible frequency range.

11:40–12:00 Panel Discussion

THURSDAY MORNING, 16 MAY 2019 JONES, 8:05 A.M. TO 12:00 NOON

Session 4aPA

Physical Acoustics and Signal Processing in Acoustics: Infrasound I

Roger M. Waxler, Cochair
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Philip Blom, Cochair
Los Alamos National Laboratory, P.O. Box 1663, Los Alamos, NM 87545

4a THU. AM

Invited Papers

8:05

4aPA1. Estimating multiple bearings-of-arrival from tornadic storms using the complex Wishart distribution. William G. Frazier, Carrick L. Talmadge, Claus Hetzer, and Roger M. Waxler (NCPA, Univ. of Mississippi, 145 Hill Dr., P.O. Box 1848, University, MS 38677, wgfrazier@gmail.com)

Several array signal processing methods can be used to estimate bearings-of-arrival (BOA) in the presence of multiple infrasound sources, and their effectiveness depends upon several factors including array geometry, relative signal and noise power spectra, and noise cross-spectra. One of the most effective and computationally efficient methods is Multiple Signal Classification (MUSIC). However, MUSIC's performance not only degrades with the decreasing signal-to-noise ratio, as all methods, but also degrades as the noise model deviates from the assumption of uncorrelated, equal noise power on all channels. Uncorrelated, but *unequal*, noise power levels are a common situation with infrasound arrays, and the degradation of MUSIC's performance has been observed when estimating 2–10 Hz acoustic emissions from tornadic storms. This presentation examines the performance of formal maximum-likelihood estimation of multiple BOAs using the complex Wishart distribution as a model for the array's cross-spectral density matrix estimates. Estimation and computational performance comparisons with MUSIC are also reported.

8:25

4aPA2. Improved infrasound array processing with robust estimators. Jordan W. Bishop, David Fee, and Curt Szuberla (Wilson Alaska Tech. Ctr., Geophysical Inst., Univ. of Alaska Fairbanks, 2156 Koyukuk Dr., P.O. Box 757320, Fairbanks, AK 99709, jwbishop2@alaska.edu)

Accurate infrasound source and path characterization rely on high-quality array processing parameter estimates. Physical and statistical assumptions underlying conventional array processing techniques sometimes fail in practice due to propagation effects or station degradation. Unlike conventional least squares regression, robust regression estimators are relatively insensitive to data that deviate from the assumed planar model. We compare two such estimators, M-estimators, and least-trimmed squares (LTS), to conventional array processing methods (frequency-wavenumber beamforming, progressive multi-channel cross correlation, L1 regression, and ordinary least squares) using synthetic and real infrasound data. Synthetic testing suggests that robust estimators are resistant to timing errors and noise contamination. We also present case studies from both International Monitoring System and the Alaska Volcano Observatory infrasound data that demonstrate how these techniques have produced accurate array processing results despite an element polarity reversal, timing error due to the loss of GPS lock, and a deviation from the plane wave assumption. We also evaluate the effectiveness of these techniques to arrays with differing geometries and number of elements, and note that the examination of LTS residuals enables outlying inter-element differential times to be flagged automatically, providing a data quality tool.

8:45

4aPA3. Using acoustic waveform inversion to quantify volcanic emissions. Alexandra M. Iezzi and David Fee (Geophysical Inst., Univ. of Alaska Fairbanks, 2156 Koyukuk Dr., UAF - Geophysical Inst., Fairbanks, AK 99775, amiezzi@alaska.edu)

Volcanic eruptions produce immense sound, particularly in the infrasound band. Acoustic waveform inversion shows promise for improved eruption characterization by providing robust estimates of erupted volume and mass. Previous inversion studies have generally assumed a simple volumetric acoustic source (monopole) that radiates sound equally in all directions. However, more complex and complete source reconstructions are possible with a combination of equivalent sources (multipole). Recent work has made progress using Finite-Difference Time-Domain modeling over high-resolution topography to obtain the full three-dimensional Green's functions. The source-time function can then be inverted for and converted to a volume and mass flow rate. We review the acoustic waveform inversion as it has been applied to volcanic eruptions and discuss current limitations and how they can be mitigated. In most cases, the simple (monopole) source mechanism is a good approximation for discrete volcanic explosions, but a small directionality (dipole) component may remain. Furthermore, the neglecting effects of topography can lead to the overestimation of both the monopole and dipole strengths. Volcano infrasound source mechanisms are also not well constrained due to infrasound sensors usually being deployed on the surface. The methods discussed here can be extended to anthropogenic explosions and monitoring efforts, potentially in near-real time.

9:05

4aPA4. Probabilistic inversion for submerged or buried source depth and strength from infrasound observations. Gil Averbuch (Dept. of GeoSci. and Eng., Delft Univ. of Technol., Graswinckelstraat 64, Delft 2613 PX, The Netherlands, g.averbuch@tudelft.nl), Roger M. Waxler (NCPA, Univ. of Mississippi, Oxford, MS), and Läslo G. Evers (R&D Dept. of Seismology and Acoust., Royal Netherlands Meteorological Inst., De Bilt, The Netherlands)

In seismology, the depth of a near surface source is hard to estimate in the absence of local stations. However, long-range infrasound propagation from an underwater or underground source is very sensitive to variations in its depth. This characteristic is employed in an infrasound based inversion for the sources depth and effective-acoustic-strength (EAS). A synthetic dataset, generated by the Fast-Field-Program (FFP), is used to investigate the accuracy of a Bayesian inversion scheme under the variations of the number of stations, source depth, and signal-to-noise ratio (SNR). SNR has proved to have the most dominant influence on the inversion precision. Results from a single station inversions with SNR = 5 had a standard deviation (SD) of $\pm$20m in depth and 10% in EAS. For SNR = 1, SD values increased to $\pm$40 m in depth and 40% in EAS. Similar results were obtained from five and ten stations inversions. This is the first attempt to extract the absolute source depth and EAS from long-range infrasound signals. Results show that infrasound may be used to accurately obtain underwater and underground source parameters.

9:25

4aPA5. Modelling ocean ambient noise with finite ocean depth and comparisons with observations. Marine De Carlo, Alexis Le Pichon (CEA/DAM/DIF, Arpajon F-91297, France, marine.decarlo@cea.fr), and Fabrice Ardhuin (Laboratoire d'Océanographie Physique et Spatiale (LOPS), Univ. Brest, CNRS, Ifremer, IRD, Brest, France)

The global International Monitoring System (IMS) network continuously detects coherent ambient infrasound noise between 0.1 and 0.5 Hz. This noise, referred to as microbaroms, is generated by the second order non-linear interaction of ocean waves, mostly during severe storms. A global and multi-year analyze of microbaroms highlights the strong influence of middle atmospheric conditions on the propagation. Various source models have been developed. Brekhovskikh *et al*. (1973) and Ardhuin and Herbers (2013) considered a source directivity effect in infinite depth ocean with radiative pressure depending on the wave elevation angle. Waxler and Gilbert (2006) and Waxler (2007) investigated the radiation of infrasound by ocean waves in finite depth ocean by monopolar sources. In this study, the combined effects of non-monopolar source and bathymetry on the radiation are addressed. Beyond theoretical issues, source modelisation and propagation through a realistic atmosphere are carried out. Comparing the predicted signals with the observed ones at all IMS stations shows good agreements for both directional and amplitude information. Building a global reference database of oceanic noise sources opens new perspectives for providing additional integrated constraints on middle atmosphere dynamics and disturbances.

9:45

4aPA6. Data-driven interpretable models of wave dynamics for infrasound monitoring. Christophe Millet (CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr) and Francois Lott (LMD, ENS, Paris, France)

Accurate and efficient models are essential to understand and predict acoustic signals propagating in the atmosphere. In many practical problems such as source localization and yield estimation, multiple models for describing the atmospheric fluctuations and infrasound are available, with varying approximation qualities. In standard practice, inferences are exercised as if the selected models had generated the observations. This approach ignores the model uncertainty, leading to biased inferences and to estimates that may be extremely sensitive to tunable parameters. This work explores a hierarchical Bayesian framework for producing interpretable models for both atmospheric gravity wave (GW) dynamics and acoustic propagation, from ground-based infrasound measurements. It is shown that there are only a few important terms that govern the GW dynamics and the interactions with infrasound. The resulting GW models can either be incorporated into global climate models to better describe the effects of GWs on the global circulation or used together with infrasound propagation models for improving inference accuracy and efficiency. This perspective, combining the resulting infrasound-driven models with sparse sensing and machine learning to monitor the atmosphere, is explored using recurring events such as the ammunition destruction explosions at Hukkakero, in northern Finland.

10:05–10:20 Break

10:20

4aPA7. Preliminary analyses of seismo-acoustic wave propagation in outdoor field-scale analog volcanic explosions. Tracianne B. Neilsen (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Robin S. Matoza, Sean Maher (Univ. of California, Santa Barbara, Santa Barbara, CA), Margaret G. McKay (Brigham Young Univ., Provo, UT), Richard Sanderson (Univ. of California, Santa Barbara, Santa Barbara, CA), Greg A. Valentine, Ingo Sonder, and Andrew G. Harp (Univ. at Buffalo, Buffalo, NY)

Shallow and subaerial volcanic processes radiate infrasound directly into the atmosphere; sampling these infrasound complement seismic data and aids with physical quantification of explosive eruption mechanisms. More advanced quantitative models of the infrasonic source and associated seismo-acoustic wave conversion and coupling have the potential to improve volcano monitoring capability. Field-scale outdoor experiments under relatively controlled conditions provide the opportunity to test, refine seismo-acoustic wave propagation and source inversion strategies, and provide a critical bridge between laboratory-scale experiments, numerical simulations, and full-scale volcano field data. We present preliminary investigations of data collected during an NSF-sponsored workshop at the University at Buffalo in July 2018. Sets of buried explosives were detonated sequentially. The explosions were recorded at 30–330 m on colocated broadband seismometers buried at 1 m, infrasound sensors, and microphones. Analyses of waveform signatures, including cross-correlation and coherence analyses, provide insights into coupling between seismic and acoustic signals over different frequency bands as a function of distance. Comparisons of the seismo-acoustic coupling for a variety of blast strengths and detonation sequences provide insights into how seismo-acoustic coupling scales with amplitude and source depth. The use of both microphones and infrasound sensors highlights the potential benefit of wideband volcano-acoustic recordings.

10:40

4aPA8. The sub-microbarom notch in acoustic wind-filter response. Thomas B. Gabrielson (Penn State Univ., P.O. Box 30, State College, PA 16804, tbg3@psu.edu)

The measurement of the frequency response of infrasound elements with spatial-averaging wind filters is often done by comparison with a reference sensor and with ambient noise as the excitation. Frequently, a notch appears in the response just below the microbarom band—a notch that is not explained by the acoustics of the wind filter. In fact, this notch is diagnostic of the spatial averaging of wind-associated turbulence. The frequency region of the notch is bounded above and below by regions in which excellent determinations of response can be made (1) below the notch under moderate- to high-wind conditions where the scale of the turbulence exceeds the scale of the wind filter rendering the wind filter ineffective and (2) in the microbarom region where the acoustic component is strong and coherent across the entire wind-filter aperture. Furthermore, the phase of the response is not affected in the region of the notch. Consequently, the true acoustic response can be estimated in the notch region in several ways. It would, however, be a mistake to ignore the information about the effectiveness of the wind filter that, in effect, creates the notch.

Contributed Papers

11:00

4aPA9. Acoustic resonances of atmospheric oscillations. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu), Nikolay A. Zabotin, and Liudmila Zabotina (Elec., Comput. and Energy Eng. Dept., Univ. of Colorado, Boulder, CO)

Atmosphere responds in a resonant way to broad-band excitation by earthquakes, volcano eruptions, and convective storms. Energetic oscillations, known as acoustic resonances, occur at frequencies of 3.5–4.5 MHz and involve infrasound propagation between lower thermosphere and either the ocean or the solid earth. Several approaches have been proposed in the literature to determine the conditions of the acoustic resonances occurrence, predict their frequencies, and relate the frequencies to thermal structure of the atmosphere. This paper presents an asymptotic theory of atmospheric resonances. Contributions to the resonance condition of the Berry phase of infrasonic waves as well as phase shifts at turning points and at reflection from the ground surface are discussed. Unlike low and middle latitudes, acoustic resonances are predicted to be a seasonal phenomenon in polar regions. Excitation of atmospheric resonances by plane-wave vertical displacements of the ground surface and by finite sources is considered. Asymptotic predictions are compared to results of numerical simulations. Infrasound tunneling between turning points via evanescent waves is shown to play a critical role in ionospheric manifestations of the acoustic resonances. [Work supported in part by NSF.]

11:15

4aPA10. Initial characterization of infrasound sources using the phase and amplitude gradient estimator method for acoustic intensity. Francisco J. Irarrazabal, Mylan R. Cook, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, firarrazabal@gmail.com), Pauline Nelson (Dept. of Phys., Brigham Young University-Idaho, Rexburg, ID), Daniel J. Novakovich, and Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The phase and amplitude gradient estimator (PAGE) method for vector acoustic intensity [Thomas *et al.*, *J. Acoust. Soc. Am.* **137**, 3366–3376 (2015)] has been used previously to improve source characterization over broad frequency ranges. This paper describes initial applications of the PAGE method to the infrasound region for outdoor sources using multi-microphone probes. Measurement challenges include wind noise, which reduces signal-to-noise ratio and coherence, low-frequency phase and amplitude mismatch between microphones, and determining an appropriate microphone spacing to maximize bandwidth. Analysis challenges include phase unwrapping above the spatial Nyquist frequency, source statistical stationarity, and balancing frequency resolution with averaging across finite record lengths. This paper discusses how these challenges are being addressed for specific sources of infrasound, namely, wind turbines and large rocket motors. [Work supported by NSF.]

11:30

4aPA11. The transition of gravity wave or mesoscale flow to boundary layer turbulence and implications for infrasound propagation conditions. Jelle D. Assink (Seismology and Acoust., Royal Netherlands Meteorological Inst. (KNMI), P.O. Box 201, De Bilt 3730 AE, The Netherlands, assink@knmi.nl) and Gregory W. Lyons (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Knowledge of the atmospheric boundary layer (ABL) is important for weather and climate forecasting as well as infrasound detection. In spite of many advances, the ABL is still not represented realistically in weather models, which leads to uncertainties in weather predictions. At the same time, the influence of the ABL on infrasound detectability is not well understood, which potentially limits the capability of the technique. In this work, a network of anemometers, microbarometers, and ceilometers is used to characterize gravity wave activity and turbulence in the ABL. The sensors are co-located at the Cabauw Experimental Site for Atmospheric Research (CESAR). The sensor network allows for the estimation of velocity and pressure spectra, as well as boundary-layer structure and mixing height. The goal of this work is to study the transition from gravity wave or mesoscale flow to boundary layer turbulence and to consider implications for infrasound propagation conditions. Here, an analysis of spectra under convective and stable conditions is presented with respect to both gravity wave and boundary-layer turbulence theory. Observed distinctions in the velocity and pressure spectral transitions will be discussed, including large-scale turbulence and the mesoscale spectral gap.

11:45

4aPA12. The effects of spatial and temporal frequency of meteorological data sampling on accurate prediction of infrasound propagation. Ross E. Alter (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, ross.e.alter@usace.army.mil), Michelle E. Swearingen (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL), Mihan H. McKenna (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS), Christopher Simpson (GeoTech. and Structures Lab., U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS), and Brian G. Quinn (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Accurately characterizing the environment is essential for robust predictions of infrasound propagation. In particular, meteorological data often exhibit large spatiotemporal variations; thus, accurate characterization can be challenging given the limitations of current measurement techniques. Though numerical weather models can provide realistic representations of the meteorological environment, the spatial and temporal sampling frequencies needed for accurate prediction of infrasound propagation are currently unknown. To address this issue, various simulations were conducted with the Weather Research and Forecasting (WRF) meteorological model, and the output was incorporated into a wide-angle parabolic equation method (PE) model to predict local infrasound propagation between 1 and 20 Hz. The sensitivity of infrasound propagation to spatial (1–15 km) and temporal ($\leq$1 min) sampling frequency within the WRF model was then calculated. Furthermore, the results of these simulations were compared to calibrated field measurements of meteorological and infrasound data at ranges up to 15 km to identify which sampling frequencies and locations along the propagation path provided the most accurate propagation results. Next steps for applying these results to future infrasound simulations will also be discussed, particularly regarding the optimization of horizontal and vertical meteorological sampling. Distribution Statement A: Approved for public release; Distribution is unlimited.

Session 4aPP

Psychological and Physiological Acoustics and Speech Communication: Spatial Hearing, Complex Acoustic Scenes, and Clinical Devices (Poster Session)

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

Contributed Papers

4aPP1. Synergy of spectral and spatial segregation cues in simulated cocktail party listening. Brianna Rodriguez (Dept. of Commun. Sci. and Disord., Univ. of South Florida-Tampa, Tampa, FL 33620, bcrodriguez@mail.usf.edu), Jungmee Lee (Commun. Sci. and Disord., Univ. of South Florida - Tampa, Madison, Wisconsin), and Robert Lutfi (Commun. Sci. and Disord., Univ. of South Florida - Tampa, Tampa, FL)

An approach is borrowed from Measurement Theory [Krantz *et al*., *Foundations of Measurement* (1971), Vol. 1] to evaluate the interaction of spectral and spatial cues in the segregation of talkers in simulated cocktail-party listening. The goal is to determine whether mathematical transformations exist whereby the combined effect of cues can be additively related to their individual effects. On each trial, the listener judged whether an interleaved sequence of 4 vowel triplets (heard over headphones) was spoken by the same $BBB_{BBB...}$ or different $ABA_{ABA...}$ talkers. The talkers had nominally different fundamental frequencies and spoke from nominally different locations (simulated using Kemar HRTFs). Natural variation in these cues was simulated by adding a small, random perturbation to the nominal values independently for each vowel on each trial. Psychometric functions (PFs) relating *d'* performance to the difference in nominal values were obtained for the cues presented individually and in combination. The results revealed a synergistic interaction of cues wherein the PFs for cues presented in combination exceeded the simple vector sum of the PFs for the cues presented individually. The results are discussed in terms of their implications for possible emergent properties of cues affecting performance in simulated cocktail-party listening. [Work supported by NIDCD R01-DC001262].

4aPP2. Context-dependent trading of binaural spatial cues in virtual reality. Travis M. Moore and G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, tmoore16@samford.edu)

A classic paradigm used to quantify the perceptual weighting of binaural spatial cues requires a listener to adjust the value of one cue, while the complementary cue is held constant. Adjustments are made until the auditory percept appears centered in the head, and the values of both cues are recorded as a *trading ratio*, most commonly in μs ITD per dB ILD. Interestingly, the existing work has shown that TRs differ according to the cue being adjusted. The current study investigated whether cue-specific adaptation-which might arise due to the continuous, alternating presentation of signals during adjustment tasks-could account for this poorly understood phenomenon. Three experiments measured TRs via adjustment and via lateralization of single targets in virtual reality (VR). Targets were 500 Hz pure tones preceded by silence or by adapting trains that held one of the cues constant. VR removed visual anchors and provided an intuitive response technique during lateralization. The pattern of results suggests that adaptation can account for cue-dependent TRs. An adaptation-based theory states that the ITD contributes most to the TR during adjustment, and adjusting the ILD results in a TR reflects contributions from both the ITD and ILD. [Work supported by NIH R01 DC016643.]

4aPP3. Perceptual weighting of elevation localization cues across frequency. Axel Ahrens (Facebook Reality Labs, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, aahr@elektro.dtu.dk) and Owen Brimijoin (Facebook Reality Labs, Redmond, WA)

Spectral cues are thought to be of particular importance in the perception of the elevation of a sound source. While some work has been done on demonstrating the importance of individual frequency bands, the relative importance of bands across a wide range of frequencies has not been firmly established. To estimate this, we built a broadband signal consisting of seven 1-ERB-wide noise bands that could each be assigned to a different elevation. The frequency range was either from 1 to 16 kHz with 3-ERB-wide spectral gaps or a higher-resolution range from 3 to 12 kHz with 1-ERB-wide spectral gaps. On each trial, each frequency band was independently convolved with a randomly chosen personalized head-related transfer function from one of seven elevations (±60 deg, 15 deg steps). In a 1-interval, 2-alternative forced choice task, listeners were asked to judge whether the sound was perceived above or below a reference stimulus presented on the horizontal plane. Two azimuth angles at -15 deg and -45 deg were considered. Perceptual weights for each frequency band were then calculated using a regression analysis method. Results showed that listeners tended to weight the 6.5 kHz band the highest for both azimuth directions and frequency resolution conditions.

4aPP4. The effect of reverberation on listening effort. Yi Shen, Yuan He, Kimberly G. Skinner, and Donghyeon Yun (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S Jordan Ave., Bloomington, IN 47405, shen2@indiana.edu)

The current study investigates whether long reverberation increases listening effort during speech recognition. Listening effort during word recognition in multi-talker babble noise was assessed with or without high levels of reverberation. A dual-task paradigm was adopted, in which the primary task was word recognition in noise at individually selected signal-to-noise

ratio (SNR) that yielded an average performance level of 50% correct, and the secondary task was a visual-tracking task with individually adjusted difficulty level to yield an average performance level of 85% correct. In each 30-s trial, seven monosyllabic words were presented sequentially at a rate of four seconds per word. Young normal-hearing listeners were instructed to verbally repeat each word while performing the secondary task. In the reverberant condition, a cascade of all-pass filters was used to achieve a reverberation time of 1 s without altering the original spectrum of the speech. For the primary task in isolation, the reverberant condition required 10 dB or more in SNR to achieve the 50% target performance level. When the listeners performed the two tasks simultaneously, no consistent adverse effect of reverberation was found on the performance of the primary or secondary task compared to the no-reverberation condition.

4aPP5. Perceptual squelch of room effect in listening to speech. Aimee Shore (Phys. and Astronomy, Michigan State Univ., 567 Wilson Rd., Biomedical Physical Sci. Bldg., MSU, East Lansing, MI 48824, shoreaim@msu.edu), Brad Rakerd (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), and William M. Hartmann (Phys. and Astronomy, Michigan State Univ., East Lansing, MI)

Room effect squelch—the auditory system's ability to suppress reverberation and coloration—has historically been entirely attributed to binaural listening. An alternative hypothesis is that a listener's own head-related transfer functions (HRTFs) are necessary for maximum squelch. Two perceptual experiments were conducted to investigate the role of individualized HRTFs. The first experiment used binaural synthesis over headphones to deliver speech stimuli to listeners. Head-related impulse responses were measured in a test room and convolved with anechoic female speech. Headphone presentation of convolved stimuli was diotic or binaural, and listeners rated the amount of perceived room effect in each stimulus. Regression analyses indicated that listeners perceived less room effect in binaural listening mode, but ratings were similar for individualized and nonindividualized HRTF conditions. Because it was thought that headphone presentation did not adequately convey HRTFs, a second experiment was conducted using loudspeakers. Transaural synthesis was used to present individualized and nonindividualized speech stimuli to listeners. Analyses indicated that listener ratings of perceived room effect were often, but not always, lower when listening to own-ear conditions. We conclude that there is limited support for the hypothesis that listeners experience maximum squelch when listening with their own ears.

4aPP6. Listening while balancing: Dual-task costs in speech vs. noise maskers. Karen S. Helfer, Richard L. Freyman (Commun. Disord., Univ. of Massachusetts Amherst, 358 N. Pleasant St., Amherst, MA 01002, khelfer@comdis.umass.edu), Richard Van Emmerik, Jacob Banks (Kinesiology, Univ. of Massachusetts Amherst, Amherst, MA), Michael Clauss, and Lincoln Dunn (Commun. Disord., Univ. of Massachusetts Amherst, Amherst, MA)

Many middle-aged adults report that listening is effortful in adverse communication situations. One means of quantifying listening effort is by measuring dual-task costs. The present study examined the influence of early aging on dual-task costs using a technique which required participants (younger and middle-aged adults) to complete a postural control task while listening to speech. For the postural control task, participants stood on a force platform and had to maintain their center of pressure within a prescribed area (denoted using real-time visual feedback). Two speech perception tasks were used, each presented with two types of maskers (same-sex two-talker speech masker and steady-state speech-shaped noise): repeating back low-predictability sentences, and listening to Connected Speech Test passages and then answering content questions based on each passage. This presentation will describe data analyses designed to uncover how listener age group and masker type influenced listening effort as measured by dual-task costs. [Work supported by NIDCD 012057.]

4aPP7. Informational masking of speech analogues by intelligible and non-intelligible but acoustically similar interferers. Robert J. Summers and Brian Roberts (Psych., School of Life and Health Sci., Aston Univ., Psych., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk)

Informational masking of target speech is generally greater when the interfering speech is intelligible than when it is not (e.g., speech from an unfamiliar language), but the relative contributions of acoustic-phonetic and linguistic interference are often difficult to assess owing to acoustic differences between interferers (e.g., different talkers). This study used three-formant analogues (F1 + F2 + F3) of natural sentences as targets and interferers. Target formants were presented monaurally (F0 = 120.3 Hz) either alone or accompanied with the contralateral ear by interfering formants from another sentence (F0 = 151.5 Hz); a target-to-masker ratio (TMR) between ears of 0, 6, or 12 dB was used. Interferers were either intelligible or rendered non-intelligible by delaying F2 and advancing F3 by 150 ms relative to F1, a manipulation designed to minimize spectro-temporal differences between corresponding interferers. Target-sentence intelligibility (keywords correct) was 67% when presented alone but fell considerably when a non-intelligible interferer was present (49%) and significantly further when the interferer was intelligible (41%). The changes in TMR produced neither a significant main effect nor an interaction with interferer type. The results suggest that although linguistic factors contribute to informational masking, interference with acoustic-phonetic processing of the target can explain much of the impact on intelligibility. [Work supported by ESRC.]

4aPP8. Is auditory distance perception in rooms binaural? Luna Prud'homme and Mathieu Lavandier (Laboratoire Génie Civil et Bâtiment, Univ Lyon, ENTPE, 3 rue Maurice Audin, Vaulx-en-Velin 69120, France, luna.prudhomme@entpe.fr)

The goal of this study was to determine whether auditory distance perception is binaural or monaural. Listeners performed an experiment in which they judged the distance of a sound source using headphones. Individualized and non-individualized binaural room impulse responses were measured to simulate sound sources placed between 1 and 4 m in front of the listener. The listening test was performed in the same room used for the measurements, and listeners were facing visual anchors. Different conditions tested the influence of controlling the sound level, individualizing the stimuli, and the amount of binaural information present in these stimuli. Results showed that binaural information does not seem to be necessary for auditory distance perception in rooms for naive listeners. However, its absence can alter externalization of sounds, which could prevent listeners from judging distance via headphones when it creates a mismatch between auditory and visual information. The variation of the sound level was a preponderant cue used by the listeners. Its absence or artificial variation greatly altered distance judgments for naive listeners.

4aPP9. Investigating the role of temporal fine structure in everyday hearing. Agudemu Borjigan and Hari M. Bharadwaj (Biomedical Eng., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, aagudemu@purdue.edu)

Human listeners can derive substantial masking release when there are discrepancies in pitch or spatial location between the target and masking sounds. While the temporal fine-structure (TFS) in low-frequency sounds can convey information about both pitch and location, a nuanced debate exists in the literature about the role of these TFS cues in masking release. The long-term goal of the present study is to leverage individual differences to understand the role of TFS in everyday hearing. As a first step, we sought to measure individual TFS sensitivity using monaural frequency modulation (FM) and binaural interaural time difference (ITD) detection tasks.

Preliminary data show large individual differences in these measures. Moreover, individual differences in ITD sensitivity were correlated with monaural FM sensitivity suggesting that monaural TFS coding can be a primary bottleneck determining binaural sensitivity. Alternately, both FM and ITD sensitivity variations could be reflecting common non-sensory factors (e.g., attention). To disambiguate between these hypotheses, we designed two passive EEG metrics of TFS coding. Follow-up experiments will compare individual differences in these perceptual and EEG measures to each other, and to speech-in-noise perception in complex environments.

4aPP10. A method for conversational signal-to-noise ratio estimation in real-world sound scenarios. Naim Mansour (Hearing Systems, Dept. of Health Technol., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kongens Lyngby 2800, Denmark, naiman@elektro.dtu.dk), Márton Marschall, Tobias May (Hearing Systems, Dept. of Health Technol., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), Adam Westermann (Widex A/S, Lynge, Denmark), and Torsten Dau (Hearing Systems, Dept. of Health Technol., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

The analysis of conversational signal-to-noise ratios (SNRs) measured in real-world scenarios can provide vital insight into people's communicative strategies and difficulties and guide development of hearing devices. However, measuring SNRs accurately and realistically is challenging in typical recording conditions, where only a mixture of sound sources is captured. This study introduces a novel method for realistic *in situ* SNR estimation, where the speech signal of a person in natural conversation is captured by a cheek-mounted microphone, adjusted for free-field conditions, and convolved with a measured impulse response to estimate the clean speech component at the receiver. A microphone near the receiver computes the noise-only component by applying a voice activity detector. Obtained SNR values are analyzed using *in situ* recordings of a real-world workspace meeting. It is shown that the temporal resolution is increased, and fluctuations in the speech level are more accurately tracked compared to a typical spectral-subtraction-derived method. The application of the proposed SNR estimation method may be valuable for compensation procedures in hearing instruments that take conversational dynamics into account.

4aPP11. Disentangling the contribution of head shadow, loudness summation, and binaural unmasking to spatial release from masking in children. Z. Ellen Peng and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53711, zpeng49@wisc.edu)

Segregating target speech from noise is crucial for children's ability to communicate effectively in everyday environments. Past research clearly shows that when target sources are spatially separated from maskers, compared with target-masker being co-located, children as young as 2–3 years old demonstrate improved speech understanding. This effect is known as spatial release from masking (SRM). Generally, studies have used free-field or dichotic listening; hence, the contributions of head shadow, loudness summation, and binaural unmasking to SRM are unknown in children. This study aimed to quantify these factors in virtual auditory space. By varying the target-masker spatial configurations (co-located versus separated) and ear conditions (monaural versus binaural), speech understanding benefit was defined as improvement in the signal-to-noise ratio to achieve an accuracy of 50%. Results from 29 children with normal hearing (6–15 years old) show that head shadow cues are dominant in providing benefit, followed by binaural unmasking. Loudness summation, through increased intensity by listening with both ears, provided little to no benefit. No age effects were found. Results also suggest a re-balancing between cues depending on listening strategies adopted by children. For example, children who relied more on binaural unmasking received less benefit for speech understanding from head shadow. [Work supported by NIH-NIDCD.]

4aPP12. Sound source localization in two-dimensions: Rotating sources and listeners. William Yost (ASU, P.O. Box 870102, Tempe, AZ 85287, william.yost@asu.edu) and M. Torben Pastore (ASU, Troy, New York)

In 1940, Wallach published the last of three articles on localizing sound sources in two dimensions: azimuth and elevation. He proposed: "Two sets of sensory data enter into the perceptual process of localization (1) the changing binaural cues and (2) the data representing the changing position of the head." Wallach explained how head motion resolves cone-of-confusion errors to support his proposal. A group of experiments in which listeners and sound sources rotated on the azimuth plane demonstrated how head motion contributes to localization in terms of both azimuth and elevation by inducing illusory perceived sound source locations. The results generally supported his proposal regarding head motion. As Wallach was not aware of aspects of current knowledge of sound source localization, such as the role of head-related-transfer function, HRTF, and the use of the resulting spectral cues, some of his conclusions turn out to be incorrect. We conducted a series of experiments similar to Wallach's to more fully examine the roles of head and sound source rotation in localizing sound sources in two-dimensional auditory space. Some of these results will be described in this presentation. [Work supported by NIDCD and Facebook Reality Labs.]

4aPP13. Audio and visual distance perception of familiar and unfamiliar objects using Wave Field Synthesis and a stereoscopic display. Sarah Richie and Jonas Braasch (Graduate Program for Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, richis@rpi.edu)

Object distance perception can be influenced both by auditory and visual cues. This work seeks to examine the influence of both perceptual domains for familiar and unfamiliar auditory and visual stimuli. For example, an alarm clock is a familiar object and a generic vibrating sphere is an unfamiliar object because the distance cannot be estimated from known dimensions. A Wave Field Synthesis (WFS) system and a stereoscopic large screen display using shutter glasses was used to create the virtual objects. Utilizing WFS allowed for sources to be placed virtually behind and in front of the speaker array. Cues were presented audio only, visual only or audio and visual simultaneously. Participants were asked for the estimated depth of the object while randomizing the above scenarios. This work expands upon a previous study [*J. Acoust. Soc. Am.* **137**, 2374] that suggested that the visual cues tend to dominate perception even when auditory cues are available. One goal of the new study is to investigate if finding holds true and if the listener is presented with more salient cues that also allow for head movements. In the previous study, the virtual environment was based on static Head-Related Transfer Functions (HRTFs).

4aPP14. Hearing impairment and reverberation preference: Results from a virtual sound space. Andrew Burleson, Kendra L. Marks, and Pamela Souza (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, andrewburleson@u.northwestern.edu)

Reverberation is regarded as a positive component of music perception and may lead to feelings of envelopment in well-designed auditoria. While relative reverberation time preferences are clear for young, normal-hearing (YNH) listeners, previous work indicates that older, hearing-impaired listeners (OHI) show less distinct preferences for reverberation time in music. OHI listeners have degraded temporal and spatial processing abilities that impact both reverberation perception and binaural processing of auditory stimuli. Previous work has been limited to earphone presentation, precluding an individualized head-related transfer function. This experiment employed these individualized auditory cues by evaluating reverberation preference in a virtual sound room for OHI and YNH listeners. Three symphonic excerpts, spatialized to simulate orchestral performance, were presented with a range of reverberation times. Listeners selected a preferred reverberation time in a series of paired comparisons. Thresholds for interaural coherence correlation (ICC)—a binaural processing measure—were obtained. Preliminary results indicate that YNH listeners have better ICC thresholds than OHI. Concordant with previous work, YNH listeners show relative reverberation time preference at roughly 2.5 s. OHI listeners show a different preference pattern than YNH. Results to date indicate that naturalistic listening cues may play an important role in music perception for OHI listeners. [Work supported by NIH.]

4aPP15. Impact of spatial variability and masker fringe on detectability of brief signal. Michelle H. Wang, Robert H. Gilkey (Psych., Wright State Univ., 3640 Colonel Glenn Hwy., Dayton, OH 45435, wang.202@wright.edu), and Brian Simpson (Air Force Res. Lab., Wright-Patterson AFB, OH)

Seemingly, there should be a close relationship between spatial release from masking and sound localization, but this is not always the case. For example, in binaural detection, randomizing the spatial parameters of the target or masker from trial to trial has little impact on threshold. In contrast, Simpson (Ph.D. dissertation, 2011) found that left/right localization judgments for a 60-ms target masked by a simultaneous 60-ms noise were considerably less accurate (equivalent to 10-dB reduction in SNR) when the location of the masker varied randomly from trial to trial than when the masker location was fixed. However, when a forward masker fringe was added, so that the noise was turned on 500 ms before the target, the impact of location variability was very small (about 1 dB, comparable to the detection literature). To determine if the presence of masker fringe could have limited the impact of spatial variability in previous detection experiments, the current study examines the effect of masker fringe and both target and masker spatial variability on detectability in conditions comparable to those of Simpson. The results will be compared to previous findings and models reported in the binaural detection and sound localization literature.

4aPP16. The effect of musical training on ecological cocktail party listening. Anneliese K. Schulz, Elissa Hoffman, and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, 901 S. Sixth St., Champaign, IL 61820, akschul2@illinois.edu)

A multitude of studies have investigated the phenomenon that experience, such as musical training, has an impact on listener performance in challenging auditory environments. Many studies examining speech-in-speech listening (i.e., the cocktail party problem) simulate an unnatural scenario where the target talker and maskers are all facing the listener. We analyzed participants' performance in a more realistic situation with a target talker facing the listener and co-located maskers with head orientations facing away from the listener (45 or 60 deg relative to the listener). We aimed to determine if musical training provided an advantage to our participants under these ecological conditions. Stimuli were presented over a loudspeaker to listeners in a sound treated booth. Preliminary data indicate that highly trained musicians (N = 6) perform better than nonmusicians (N = 25) in our task. Musical training may improve auditory functioning in challenging ecological listening situations. Data collection for listeners with extensive musical training is ongoing.

4aPP17. Efficacy of iPad "spatial release" application. Allison Holtz, Kelli Clark, and Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, aholtz3@students.towson.edu)

Spatial Release from Masking (SRM) is the ability to obtain better speech recognition thresholds when the maskers are spatially separated from the target. Here, we present SRM data collected using three techniques: over headphones using a virtual speaker array, using Spatial Release iPad application (https://bgc.ucr.edu/games/spatialrelease/), and loudspeaker presentation in a sound-attenuated room. For all three techniques, Coordinate Response Measure (CRM) sentences were used as the stimuli, and "Charlie" was the call sign. A progressive tracking procedure was used to estimate the Speech Recognition Thresholds (SRTs) for listeners with varying hearing thresholds. The target sentence was always presentedg at 0 deg azimuth angle whereas the maskers were colocated (0 deg) with the target or symmetrically spatially separated by ±15 deg, ±30 deg, or ±45 de. Initial data analysis revealed similar SRTs for the iPad and headphone conditions and slightly poorer thresholds for the loudspeaker array condition. This was true for all spatial separations between the target and the maskers. The individual effects of age and hearing loss on spatial release from masking will be discussed. These data will aid clinicians to rapidly characterize difficulties perceived by individuals in everyday listening scenarios and to evaluate patient progress with hearing aid adjustments and aural rehabilitation over time.

4aPP18. A deep learning based segregation algorithm to increase speech intelligibility for hearing-impaired listeners in reverberant-noisy conditions. Yan Zhao, DeLiang Wang (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH), Eric Johnson, and Eric Healy (Speech & Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu)

Recently, deep learning based speech segregation has been shown to improve human speech intelligibility in noisy environments. However, one important factor not yet considered is room reverberation, which characterizes typical daily environments. The combination of reverberation and background noise can severely degrade speech intelligibility for hearing-impaired (HI) listeners. In the current study, a deep learning based time-frequency masking algorithm was proposed to address both room reverberation and background noise. Specifically, a deep neural network was trained to estimate the ideal ratio mask, where anechoic-clean speech was considered as the desired signal. Intelligibility testing was conducted under reverberant-noisy conditions with reverberation time T60 = 0.6 s, plus speech-shaped noise or babble noise at various signal-to-noise ratios. The experiments demonstrated that substantial speech intelligibility improvements were obtained for HI listeners. The algorithm was also somewhat beneficial for normal-hearing (NH) listeners. In addition, sentence intelligibility scores for HI listeners with algorithm processing approached or matched those of young-adult NH listeners without processing. The current study represents a step toward deploying deep learning algorithms to help the speech understanding of HI listeners in everyday conditions. [Work supported by NIH.]

4aPP19. A deep learning algorithm to increase intelligibility for hearing-impaired listeners in the presence of a competing talker and reverberation. Eric Healy (Speech & Hearing Sci., The Ohio State Univ., Columbus, OH), Masood Delfarah (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH), Eric Johnson (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, johnson.7289@buckeyemail.osu.edu), and DeLiang Wang (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH)

For deep learning based speech segregation to have translational significance as a noise-reduction tool, it must perform in a wide variety of acoustic environments. In the current study, performance was examined when target speech was subjected to interference from a single talker and room reverberation. Conditions were compared in which an algorithm was trained to remove both reverberation and interfering speech, or only interfering speech. A recurrent neural network (RNN) incorporating bidirectional long short-term memory (BLSTM) was trained to estimate the ideal ratio mask (IRM) corresponding to target speech. Substantial intelligibility improvements were found for hearing-impaired (HI) and normal-hearing (NH) listeners across a range of target-to-interferer ratios (TIRs). HI listeners performed better with reverberation removed, whereas NH listeners demonstrated no preference. Algorithm benefit averaged 56% points for the HI listeners at the least-favorable TIR, allowing these listeners to numerically exceed the performance of young NH listeners without processing. The current study highlights the difficulty associated with perceiving speech in reverberant-noisy environments, and it extends the range of environments in which deep learning based speech segregation can be effectively applied. This increasingly wide array of environments includes not only a variety of background noises and interfering speech but also room reverberation. [Work supported by NIH.]

4aPP20. Integration of auditory and tactile stimuli in the perception of building noise and vibration. Ben Loshin and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, loshib@rpi.edu)

Auralization is used in architectural and environmental planning to build a visceral understanding of a design. However, current auralization techniques are limited to the auditory sensory modality, while real environments require the integration of complex stimuli across multiple modalities. This study explores the human perception of coupled sound and vibration sources encountered in real spaces through the creation of immersive virtual

representations of those spaces. Acoustic and vibration responses are combined with common building noises and simulated on a calibrated motion platform, incorporating vertical whole-body vibration with binaural audio. Test participants are asked to make judgments of relative loudness and annoyance of building sounds simulated in combination with different vibrational contents. Test results are compared with those published in other studies on the psychophysics of audio-tactile summation, and the implications of the results are discussed with respect to the perception of building noise.

4aPP21. Effects of age and hearing loss on spatial release from speech-on-speech masking, performance in envelope-based psychophysical tasks, and EEG envelope-following responses. Chhayakanta Patro (Psych., Univ. of Minnesota, 311 Harvard St. SE, Apt. 805, Monneapolis, MN 55414, cpatro@umn.edu), Alix Klang (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Heather A. Kreft (Psych., Univ. of Minnesota, Minneapolis, MN), and Magdalena Wojtczak (Psych., Univ. of Minnesota, New Brighton, MN)

Behavioral measures of amplitude modulation (AM) detection and envelope interaural-phase-difference (eIPD) detection reflect listeners' ability to process temporal information. Robust encoding of temporal envelopes is necessary for understanding speech in a complex acoustic environment and for spatial segregation of a target speech from interfering background. It has been suggested that a large variability in performance in psychophysical tasks involving temporal envelope processing and in spatial release from masking for speech intelligibility may arise from cochlear synaptopathy. However, many studies have not found significant correlations between these measures and the amount of self-reported noise exposure in young listeners with audiometrically normal hearing. Similarly, electroencephalographic envelope-following responses did not significantly correlate with noise exposure or with behavioral performance reliant on envelope processing young normal-hearing population. In this study, behavioral measures in psychophysical tasks (AM and eIPD detection) and speech intelligibility in two-talker babble were measured for listeners with normal and near-normal hearing across a wide age range (20 to 69 years). Correlational analyses were performed using behavioral measures and envelope-following responses collected from the same listeners. Results will be discussed in terms of sensitivity of these measures to effects of aging and high-frequency hearing loss. [Work supported by NIH Grant R01 DC015987.]

4aPP22. Reverberation detection threshold estimates in normal-hearing listeners. Pavel Zahorik (Dept. of Otolaryngol. and Communicative Disord., Heuser Hearing Inst. and Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu) and James Shehorn (Heuser Hearing Inst., Louisville, Arizona)

The study of human sensitivity to a single acoustic reflection (echo) has a long and rich history. The influence of time delay, level, direction, and source material are well documented. Unfortunately, real world listening seldom involves only a single reflection. Multiple reflections and reverberation are instead the norm. It is therefore surprising that the detection threshold for acoustical room effects (early reflections plus reverberation) has not been extensively studied, if at all. This study represents an initial step to fill this gap in knowledge. Using virtual auditory space techniques to simulate room acoustic sounds fields over headphones, the detection threshold for reflected/reverberant sound energy was measured for three sound field conditions: a small office-sized room (broadband T60 = 0.5 s), a concert hall (broadband T60 = 1.5 s), and a reference condition with a single echo at 40 degrees to the right of midline. The source signal was a 220 Hz complex tone, 250 ms in duration. Thresholds for the single-echo reference condition and the small room condition were found to be comparable, whereas the concert hall produced thresholds that were at least 20 dB lower. Temporal integration and binaural effects are considered as potential explanations for these results.

4aPP23. Binaural modeling from an evolving habitat perspective. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Functional binaural models have been used since the mid-20th century to simulate laboratory experiments. The goal of this chapter is to extend the capabilities of a cross-correlation model so it can demonstrate human listening in complex scenarios found in nature and human-built environments. A ray-tracing model is introduced that simulates a number of environments for this study. This chapter discusses how the auditory system is used to read and understand the environment and how tasks that require binaural hearing may have evolved over the course of human history. As use cases, sound localization in a forest is examined, as well as the binaural analysis of spatially diffuse and rectangular rooms. The model is also used to simulate binaural hearing during a walk through a simulated office-suite environment. [Work supported by NSF BCS-1539276 and CISL.]

4aPP24. Relationship between localization acuity and spatial release from masking. Nirmal Kumar Srinivasan and Jess Wince (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu)

Spatial Release from Masking (SRM) is defined as the ability to obtain better Speech Recognition Thresholds (SRTs) when the masking sounds are spatially separated from the target sound. Localization refers our ability identify the direction of the sound and localization acuity is measured as the difference in locations between the actual and perceived locations. SRM and localization share many common cues on how the task is performed. Here, we present data from young normal hearing on SRM task using Coordinate Response Measure (CRM) sentences and localization acuity using three different noise Gaussian white noise bursts: low pass (1/3 octave wide centered at 500 Hz), high pass (1/3 octave wide centered at 3150 Hz), and broadband (200–5000 Hz). Thirteen loudspeakers (Orb Mod 1), separated by 15 deg in the frontal plane were used to present the stimuli. Initial analyses of the results indicated that, as expected, all the listeners obtained substantial spatial release from masking consistent with the literature, filtering the broadband noise had little effect on localization acuity for listeners with normal hearing. Finally, the relationship between SRM and localization acuity will be discussed.

4aPP25. Comparison of localization assessments for hearing protection. Eugene Brandewie (Dept. of Psych., GN Adv. Sci., 75 E. River Rd., Minneapolis, MN 55455, ebrandewie@gnresound.com) and Andrew Dittberner (GN Adv. Sci., Glenview, Illinois)

Sound localization is a critical component of the assessment of situational awareness for hearing protection devices (HPDs). A new standard is being developed for the assessment of sound localization in the horizontal-plane concerning head-worn devices. One of the methodologies offered by this standard is a quick and easy-to-use paradigm for rapid prototyping of hearing devices. This study compared the results of this new assessment methodology with a more traditional assessment of horizontal-plane sound localization. Four devices (including open-ear) were tested on normal-hearing individuals using both paradigms. A comparison of localization errors, front-back-reversals, hardware for data collection set-ups, and expected data collection time is presented. Results suggest comparable differences between the two methodologies.

4aPP26. Difference limens for noise bandwidth discrimination in listeners with normal and impaired hearing. Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, alexan14@purdue.edu)

Frequency-lowering (FL) in hearing aids is often used to move inaudible high-frequency energy from sibilant fricatives to spectral regions where hearing better. Clinically, FL settings that maximize the spectral separation

between these sounds, which can be modeled as bands of noise, are assumed to maximize discrimination between them and other speech contrasts. The purpose of this study was to quantify the minimum spectral differences for normal-hearing and hearing-impaired listeners to discriminate between bands of frozen noise. Noise bands corresponded to the average frequency and bandwidth of sibilant fricatives after undergoing FL using settings appropriate for mild-to-moderate, moderately-severe, and severely-profound hearing losses. Noise bands differed on the low- and/or the high-frequency edges. Discrimination in normal-hearing listeners was constant across presentation level and the three frequency ranges. Discrimination was also better for high-frequency edge differences. Neural excitation patterns generated from an auditory nerve model account for these findings. Neural excitation patterns generated for the three severities of hearing loss indicate that hearing-impaired listeners will rely heavily on spectral differences on the low-frequency edge and indicate that sibilant fricatives processed with FL will not be able to be discriminated solely on the basis of high-frequency edge differences. [Grant supported by Sonova USA, Inc.]

4aPP27. Can listeners reliably identify their preferred amplification profiles for speech listening? Donghyeon Yun, Yi Shen, and Zhuohuang Zhang (Speech and hearing Sci., Indiana Univ. Bloomington, 1603 E. 3rd St. 216, Bloomington, IN 47401, dongyun@iu.edu)

Personal hearing devices, such as hearing aids, may be fine-tuned for individual users' preferences by allowing them to self-adjust the amplification profiles. The purpose of the current study was to compare two self-adjustment methods in terms of their test-retest reliability. Both methods estimated preferred amplification profiles in six octave-frequency bands using the method of adjustment. In one method (method A), listeners adjusted the gain in one of six frequency bands using a programmable knob on a given trial; while in the other method (method B), listeners adjusted the gains in all six bands simultaneously according to a linear model using the same programmable knob. Ten normal-hearing listeners participated in the study. The experiment was completed in two test sessions, at least one week apart. During each session, the preferred amplification profile was estimated using both methods. Running speech in quiet or in speech-shaped noise was used as the test stimuli. At the beginning of each method, the initial amplification profile was generated randomly with the gains drawn from a uniform distribution spanning between -25 and 25 dB. The test-retest reliability for method B was better than method A. For method B, the test-retest reliability was better at lower signal-to-noise ratios.

4aPP28. Perception of musical instruments and music genres in cochlear implant recipients. Ying Hsiao, Valeriy Shafiro, Chad Walker, Jasper Oh, Megan Hebb, Kelly Brown, Stanley Sheft (Dept. of Commun. Disord. and Sci., Rush Univ., 600 S. Paulina St., Chicago, IL 60612, ying_y_hsiao@rush.edu), Kara Vasil, and Aaron C. Moberly (Dept. of Otolaryngol.-Head & Neck Surgery, Ohio State Univ. Wexner Medical Ctr., Columbus, OH)

We examined perception of musical and nonmusical stimuli in 17 experienced postlingually deafened cochlear implant (CI) recipients and 10 normal-hearing (NH) listeners using real-world music excerpts derived from the Appreciation of Music in Cochlear Implantees (AMICI) test. Following stimulus presentation, participants selected one most appropriate option among nine instrument or five genre options. Compared to NH, CI listeners demonstrated reduced instrument (99.1% vs. 68.7%) and genre (96% vs. 55.7%) identification performance. For CI listeners, the least accurately identified instruments were flute (17.6%) and saxophone (37.2%), while the drums were most accurately identified (98%). The flute was most often confused with strings (76.5% error) and the saxophone was most confused with brass instruments (23.5% error). The least accurately identified genres were Latin (41.2%) and Rock "n" Roll (41.2%), while Classical was most accurately identified (82.5%). Latin was most often confused with Rock "n" Roll (26.2% error), and Rock "n" Roll was most often confused with Country (28.7% error). For the CI recipients, instrument and genre identification strongly and significantly correlated with recognition of environmental sounds, sentences in noise, and frequency pattern discrimination. These results indicate considerable deficits in music perception in CI recipients and indicate the need for further rehabilitation. Correlations with speech further suggest a potential for cross-domain improvements.

4aPP29. 3D printed pinna embedded in circumaural hearing devices for spectral cue preservation. Carlos Acosta Carrasco (W.M. Keck Ctr. for 3D Innovation, The Univ. of Texas at El Paso, 500 W University Ave., El Paso, TX 79968, cfacostacarrasco@miners.utep.edu), Vidya Krull, Andrew Dittberner (GN Adv. Sci., GN Hearing, Glenview, IL), and Ryan Wicker (W.M. Keck Ctr. for 3D Innovation, The Univ. of Texas at El Paso, El Paso, TX)

Innovations in additive manufacturing [three-dimensional (3D) printing] have allowed for the fabrication of objects as complex as the human ear. The visible part of the human outer ear (pinna) serves as a funnel and a natural filter for incoming sound. Spectral cues generated by the pinna help with auditory localization and externalization. In an attempt to preserve spectral cues when using circumaural hearing devices, the present work explored the use of 3D printing to fabricate individualized pinna within custom-designed and fabricated hearing devices. Through 3D scanning, a computer aided design (CAD) model of a pinna from an anthropometric mannequin was generated to replicate human pinnae. Multiple 3D printing technologies were used to fabricate the CAD model, investigating different material options, dimensional accuracies, and overall printing costs. The fabricated pinnae were subjected to acoustic testing to assess spectral cue preservation by comparing mannequin head-related transfer functions obtained with the printed pinnae to those with the original. A sample circumaural hearing device with the pinna embedded within it was then designed, fabricated, and subjected to the same acoustic testing for comparison. Results from testing will be described within the context of providing individualized circumaural hearing devices that assist with spectral cue preservation.

4aPP30. Exploring the relationship between sound localization and individual use of spectral and temporal cues among hearing-impaired listeners. Gregory M. Ellis and Pamela Souza (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL 60201, gregory.ellis@northwestern.edu)

When locating a sound source, listeners are expected to rely more on spectral cues for elevation and temporal cues for azimuth. This work investigates whether sound localization ability can be predicted by individually-measured use of spectral and temporal cues. Participants were older adults with sensorineural hearing loss. Sound sources for the localization task were created in a virtual room, using a mix of virtual and physical loudspeakers in the front hemifield of the listener. Sources were distributed evenly between ± 90 deg azimuth and ± 20 deg elevation. The signal was a 1 s broadband 4-Hz amplitude-modulated noise. To assess the relationship between localization ability and use of spectral and temporal cues, listeners performed cue weighting and cue discrimination tasks. In the cue weighting task, listeners identified synthetic speech sounds that varied spectro-temporally. The cue discrimination task measured the smallest detectable difference in either spectral or temporal information among the same set of ambiguous speech sounds. Together, these tasks form a cue profile which identifies whether the listener relies to a greater extent on temporal or spectral cues. Older hearing-impaired listeners varied in their ability to localize sounds. Localization results will be discussed in the context of cue profiles and audiometry. [Work supported by NIH.]

4aPP31. Benefits from different types of acoustic beamforming in bilateral cochlear-implant listeners. David Yun (Hearing and Speech Sci., Univ. of Maryland, College Park, 0100 Lefrak Hall, College Park, College Park, MD 20742, davidyunjr@gmail.com), Todd R. Jennings, Christine Mason, Gerald Kidd (Dept. of Speech, Lang. & Hearing Sci., Boston Univ., Boston, MA), and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, College Park, MD)

Acoustic beamforming improves speech reception in "noise" at the cost of removing spatial cues. Recently, novel approaches have been proposed for enhancing speech reception via beamforming while preserving sound localization by reintroducing natural binaural cues to the beamformer output. It is, however, unclear whether such a hybrid approach will improve

performance for bilateral cochlear-implant (BICI) listeners because they receive little benefit from natural spatial cues. Here, effects of beamforming on masked speech reception thresholds (SRTs) were determined in BICI and normal-hearing (NH) listeners for a standard single-channel beamformer and a new triple-beamformer comprising a diotic central beam, -40 deg left-ear-only beam, and +40 deg right-ear-only beam [Jennings and Kidd, *JASA* **143**, 1743]. Speech targets and maskers were co-located (0 deg) or maskers were symmetrically separated at ±30 deg or ±90 deg. Natural binaural listening, single-channel beamformer, and triple-beamformer conditions were tested. Both beamformers were better than natural cues. For BICI listeners, single-channel beamforming improved SRTs by 5 (±30 deg) and 15 dB (±90 deg). SRTs for the triple-beam were best for NH but were 5 dB worse than the single-channel beam for BICI listeners. These preliminary findings suggest beamforming provides significant spatial release from masking for BICI listeners but assessing overall benefit, including localization, awaits further study.

4aPP32. Selective deficits in a clinical population with self-reported hearing difficulties but normal audiometric thresholds. James Shehorn (Heuser Hearing Inst., 117 E Kentucky St., Louisville, KY 40203, jshehorn@thehearinginstitute.org), Olaf Strelcyk (Sonova U.S. Corporate Services, Cincinnati, OH), and Pavel Zahorik (Otolaryngol. Head and Neck Surgery and Communicative Disord., Univ. of Louisville, Louisville, KY)

Audiologists often encounter patients who report hearing difficulties despite having normal audiometric thresholds. Many of these patients are told that they have normal hearing, although it is possible that the current typical audiometric test battery is not sensitive to these hearing difficulties. A test battery including immittance testing, binaural listening tasks, cognitive testing and subjective questionnaires of speech understanding, spatial hearing, and annual noise exposure was administered to 26 patients (age range: 18–53 years) who had sought out audiologic assessment at a community clinic but had normal hearing sensitivity. Despite normal hearing sensitivity, the participants in the study exhibited deficits in several binaural listening tasks, weakened middle-ear muscle reflexes, and rated their speech understanding and spatial hearing as being significantly worse than a normative population with normal hearing sensitivity. This patient group did not perform significantly worse on any of the cognitive measures nor did they report significantly more noise exposure than a normative young adult population; however, nearly two-thirds of participants were classified as "high risk" for noise exposure. These findings confirm self-reported hearing difficulties reported by this audiometrically "normal" population and suggest that several of the measures used in this study should be considered for standard audiologic evaluation.

4aPP33. Portable psychoacoustics with passive and active noise-attenuating headphones. Esteban S. Lelo de Larrea Mancera (Hearing & Speech Sci., Univ. of Maryland, 900 University Ave., Psych. Bldg. Rm. 3209, Riverside, California 92521, elelo001@ucr.edu), Trevor Stavropoulos (Psych., Univ. of California Riverside, Riverside, CA), Frederick J. Gallun (Dept. of Veterans Affairs, VA RR&D NCRAR, Portland, OR), David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL), Eric C. Hoover (Hearing & Speech Sci., Univ. of Maryland, Tampa, Florida), and Aaron Seitz (Psych., Univ. of California Riverside, Riverside, CA)

This presentation will describe data collected using a freely available procedure for psychoacoustical testing that harnesses commercially available tablet computer technology to address the current gap between (1) modern auditory neuroscience and psychophysics and (2) current clinically available tests of hearing. Portable Automated Rapid Testing (PART) measures running on an iPad were used to evaluate the detection of (a) tones in noise; (b) spectral, temporal, and spectro-temporal modulation; (c) monaural and binaural frequency modulation; and (d) brief temporal gaps inserted between brief tone pulses. Listeners also performed a spatial release from speech-on-speech masking task. Data from 151 UCR undergraduates were collected using both passive and active noise-attenuating headphones in a quiet environment and in the presence of recorded cafeteria noise. Across these and several other manipulations of equipment and threshold-estimation techniques, performance approximated that reported in the literature. These data provide a distribution of thresholds that can now be used as a normative baseline against which auditory dysfunction can be identified in future work.

4aPP34. Musical emotion perception in bimodal patients: Relationship between bimodal benefit and neural representation of temporal fine structure using Rhodes piano stimuli. Kristen D'Onofrio (Hearing and Speech Sci., Vanderbilt Univ., 1215, Nashville, TN 37215, kristen.l.donofrio@vanderbilt.edu), Spencer Smith (Univ. of Texas at Austin, Austin, TX), David Kessler, Grace Williams, and René Gifford (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN)

Combining electric and acoustic hearing across ears allows significant "bimodal hearing" benefit for speech recognition, sound quality, and music perception. The degree of bimodal benefit for speech recognition and musical emotion perception is significantly correlated with neural representation of F0 envelope using the frequency following response (FFR) for a 170-ms /da/ stimulus (D'Onofrio *et al*., in prep). The purpose of the current study is to examine the relationship between bimodal benefit for musical emotion perception and neural representation of F0 using Rhodes piano stimuli at the following fundamental frequencies: 98 Hz (G2), 262 Hz (C4), and 440 Hz (A4). Our hypotheses are (1) the correlation between bimodal benefit and neural representation of F0 and temporal fine structure will be strengthened via use of a "music" stimulus, compared to the /da/ "speech" stimulus, and (2) bimodal benefit for speech recognition will be better explained via FFR for speech stimuli. Stimuli were presented at 90 dB SPL to the non-implanted ear of bimodal listeners using magnetically shielded insert earphones. Implications regarding the clinical utility of FFR will be discussed, with particular attention given to its use as an objective measure of expected bimodal benefit for speech recognition and musical emotion perception.

4aPP35. Spatial release from masking and sound localization using real-time sensorineural hearing loss and cochlear implant simulation. Hannah M. Wright, Wesley Bulla, and Eric W. Tarr (Audio Eng., Belmont, 1900 Belmont Boulevard, Nashville, TN 37212, hannah.wrightmusic@gmail.com)

Simulations of sensorineural hearing loss (SNHL) and bilateral cochlear implantation (BCI) have been modeled successfully under static and non-real time conditions. This study performed two experiments testing the validity of a novel real-time SNHL/BCI simulation application for iOS using an in-ear binaural-recording headphone apparatus. The first experiment measured spatial release from masking (SRM) with normal hearing (NH), against headset apparatus simulations of NH, SNHL, and BCI using HINT sentences, speech shaped noise and forward masking. A one-sample *t-test* revealed significant differences between NH and simulated SNHL and BCI conditions showing reduced benefit from SRM. The second experiment employed noise bursts across nine frontal-plane loudspeakers and measured localization accuracy under NH and six frequency band BCI simulation conditions. Repeated measures two-way ANOVA and Cronbach's Alpha suggested significantly reduced localization ability with BCI simulation. While further testing is needed, results here provide promising evidence that real-time binaural recording with low-latency processing and in-ear playback may be used to simulate SNHL and the BCI percept in NH listeners. The limitations and potential of this technology to expand the subject pool and expedite innovative testing are discussed.

4aPP36. Predicting speech-cue weighting in older people with impaired hearing. Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Richard Wright (Dept. of Linguist, Univ. of Washington, Seattle, WA), and Pamela Souza (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Previously published data (Souza *et al*., 2015; 2018) revealed individualized patterns of cue-weighting in the identification of synthetic speech (bah, dah, lah, wah). The speech tokens were constructed such that temporal modulation (TM; envelope rise-time) and spectrotemporal modulation (STM; formant transitions) were systematically varied. Here, a

discrimination task with the speech stimuli and a modulation detection task with noise tokens were related to cue weighting for thirty people aged 61–90 years with impaired hearing. Discrimination of STM in the speech stimuli was correlated with both STM sensitivity and cue weighting, while TM discrimination and the audiogram were not. The direct link between STM sensitivity and cue weighting was not supported, however. Results suggest that there was no substitute for actually measuring the ability to discriminate the speech stimuli on the basis of spectrotemporal information. The discrimination task may provide a more time-efficient measure of cue weighting than the cue identification task. These results have important implications for the rehabilitation of impaired hearing, which is generally guided only by the audiogram, which was the worst predictor of cue weighting tested. [Work supported by NIH R01 DC006014 (PI: Souza) and NIH R01 DC015051 (PI: Gallun).]

4aPP37. Effects of training on sensitivities to spatial changes in auditory scenes. Natalie J. Ball (Cognit. Psych., Univ. at Buffalo, 373H Park Hall, University at Buffalo, Buffalo, NY 14260, njball@buffalo.edu), Matthew Wisniewski (Kansas State Univ., Manhattan, KS), Brian Simpson (U.S. Air Force Res. Lab., Wright-Patterson AFB, OH), and Eduardo Mercado (Cognit. Psych., Univ. at Buffalo, Buffalo, NY)

Previous work has shown that change deafness can occur with changes in the spatial location of objects within auditory scenes. Whether performance can be improved with training has yet to be directly tested. In the present study, the impact of training was examined in a "flicker"-like paradigm, where an initial scene comprising environmental sounds presented on the horizontal plane alternated with the presentation of a comparison scene, which was either the same or contained a change in the location of one or more sounds. Trained participants were trained on a set of sounds on day 1, while control subjects completed an unrelated visual task. On day 2, participants were tested using the same paradigm, with trained subjects hearing either the sounds they were trained on or new sounds, and controls being exposed to the paradigm for the first time. Overall, trained participants performed better (93% accuracy) than untrained participants (79%). Both groups had lower reaction times on correct-response change trials than on correct-response no-change trials. Trained subjects performed no better on trained sounds than new sounds. These data indicate that training can improve task performance, but improvements may be not be limited to sounds experienced during training.

4aPP38. Further analysis of behavioral measures of cochlear gain and gain reduction in listeners with normal hearing or minimal cochlear hearing loss. Elizabeth A. Strickland, Miranda Skaggs, Nicole Mielnicki, William Salloom, Hayley Morris, and Alexis Holt (Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, estrick@purdue.edu)

This is a continuation of a study examining the relationship between cochlear hearing loss and psychoacoustic measures thought to be related to a cochlear function. In the listeners tested, audiometric thresholds for long tones ranged from well within the clinically normal range to just outside this range. Where thresholds were elevated, other clinical tests were consistent with a cochlear origin. Because the medial olivocochlear reflex decreases cochlear gain in response to sound, when possible, measures were made with short stimuli. Signal frequencies were from 1 to 8 kHz. One point on the lower leg of the input/output function was measured by finding threshold masker level for a masker almost one octave below the signal frequency needed to mask a signal at 5 dB SL. Gain reduction was estimated by presenting a pink broadband noise (BBN) precursor before the signal and masker and measuring the change in signal threshold as a function of the precursor level. Previous studies with listeners with normal hearing have shown that gain reduction begins at a low precursor level and grows compressively as the precursor level is increased. The current study is designed to determine whether this pattern changes when cochlear gain is permanently reduced. [Work supported by NIH(NIDCD)R01 DC008327 (EAS), the Purdue Office of the Executive Vice President for Research, and the Purdue Graduate School (WBS).]

4aPP39. Effects of self-reported hearing difficulty on intensity discrimination judgments. Gwen O. Saccocia and Joseph C. Toscano (Dept. of Psychol. and Brain Sci., Villanova Univ., 800 E Lancaster Ave., Villanova, PA 19085, gsaccoci@villanova.edu)

Listeners may report difficulty understanding speech, particularly in background noise, despite having normal audiograms that do not suggest a sensorineural hearing loss. One potential cause of this hearing difficulty is auditory neuropathy (AN), a disruption in the function of the auditory nerve. AN may specifically affect auditory nerve fibers that code intensity differences at higher sound levels, resulting in particular difficulty with speech recognition but preserved audiometric thresholds. This study aims to detect AN by measuring intensity discrimination thresholds at different frequencies and sound levels, comparing performance to self-reported measures of hearing difficulty. Listeners performed an intensity discrimination task where they heard pairs of tones and judged whether the first or second tone was louder. Psychophysical functions were computed to measure listeners' discrimination thresholds and point of subjective equality (PSE, i.e., point at which the two tones are judged to have equal intensity). Results showed that listeners who report greater speech-in-noise difficulty had shifted PSEs specifically at higher sound levels (60–70 dB SPL, the range used in conversational speech), such that they were more likely to perceive the second tone as louder than it was. The results suggest that an intensity discrimination task may be a useful test for AN.

4aPP40. Experiment on the relationship between the dynamic range of hearing and psychophysical tuning curves. Marc Brennan (Special Education and Commun. Disord., Univ. of Nebraska-Lincoln, 4075 East Campus Loop South, Lincoln, NE 68583, Marc.Brennan@unl.edu)

In this study, we hypothesized that improving the dynamic range of hearing for adults with hearing loss would improve spectral resolution by providing better audibility across frequency and—due to the higher sensation level—improved phase locking. However, limited outer hair cell (OHC) function and spread of excitation likely curtail the benefit of improved audibility on spectral resolution. In this experiment, the relationship of the dynamic range of hearing to measures of psychophysical tuning curves was quantified for 13 adults with hearing loss. Twenty-one adults with normal hearing served as controls. A dynamic range of hearing was manipulated by systematically adjusting hearing-aid gain and compression ratios. To better understand the mechanisms that support the encoding of spectral resolution with amplification, the behavioral data were modeled using a cochlear excitation model. For most listeners, the psychophysical tuning curves were similar regardless of changes in the dynamic range of hearing. For a subset of participants, tuning on the low frequency side improved as the dynamic range of hearing was increased—which could be accounted for in the model by residual OHC function. The model better accounted for unaided than aided measures of tuning.

4aPP41. A software controlled audiometric setup to predict types of hearing loss from types of response delays. Amitava Biswas, Edward L. Goshorn, and Jennifer Goshorn (Speech and Hearing Sci., Univ. of Southern Mississippi, 118 College Dr. #5092, SHS Res. Lab., Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Diagnosis of certain types of non-organic hearing loss is often confused with organic hearing loss. A software controlled audiometric setup has been designed by the authors for this purpose. Its salient features are (1) partially randomized step sizes, (2) partially uncorrelated increase and decrease in the stimulus intensity after each response or no response of the subject, (3) recording of corresponding response delays of the subject, and (4) providing audio-visual feedback. Its sensitivity, specificity, and applications in clinics and the industry will be discussed.

4aPP42. Effects of auditory-nerve loss on tone detection in roving-level noise. Kenneth S. Henry, Kassidy N. Amburgey (Otolaryngol., Univ. of Rochester, 601 Elmwood Ave., Box 629, Rochester, NY 14642, kenneth_henry@urmc.rochester.edu), and Kristina S. Abrams (Neurosci., Univ. of Rochester, Rochester, NY)

Auditory-nerve (AN) loss has emerged as a significant public health concern because it occurs steadily with age and potentially following noise-induced temporary threshold shifts. AN loss without hair-cell damage remains undetectable with an audiogram yet is commonly assumed to degrade auditory perception under real-world, noisy conditions. Here, we tested whether AN loss impacts behavioral tone-in-noise (TIN) detection in the budgerigar, an avian species with sensitivity similar to humans on many simple and complex listening tasks. AN damage was induced with kainic acid and confirmed using auditory evoked potentials and otoacoustic emissions. TIN thresholds were quantified in 1/3-octave noise as a function of frequency and sound level using operant conditioning and two-down, one-up, adaptive tracking procedures. Kainic acid reduced gross AN potentials by 40%–70% across animals without impacting otoacoustic emissions. TIN thresholds in control animals decreased with increasing frequency and showed minimal elevation ($<$1 dB) when sound level was roved $\pm$10 dB across trials. TIN thresholds in kainic-acid exposed animals were as sensitive as in the control group and showed similar preservation with roving sound level. These results suggest a minimal impact of AN loss on behavioral TIN detection, even under conditions requiring rapid adaptation to changing sound level.

4aPP43. Evaluating new hearing aid technologies in laboratory simulations of listening scenarios. Peggy B. Nelson, Elizabeth Anderson, Trevor T. Perry, Kristi Oeding, and Andrew Byrne (Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu)

It can be important for clinical researchers to be able to evaluate the performance of sensory aids using both objective and subjective methods. New technology (such as self-fit hearing aids) can be evaluated in a laboratory setting in calibrated listening scenarios that reflect daily listening situations. In the Center for Applied and Translational Sensory Science (CATSS) multisensory laboratory, we have developed simulations of challenging conversational scenarios so that users of sensory aids can make judgments of sensory aid performance in realistic but controlled conditions. Listeners with hearing loss make ratings of intelligibility, sound quality, and preference in scenarios such as small group conversations and entertainment listening. At the same time, measures of hearing-aid gain and speech intelligibility are obtained. These ratings are compared to outcome measures such as the Speech, Spatial, and Qualities of Hearing Scale (SSQ; Gatehouse and Noble, 2004) and Social Participation restrictions questionnaire (SPaRQ; Heffernan *et al*., 2018) to determine relationships between intelligibility, preference, benefit, and hearing aid gain. Results will help refine methods for evaluating the performance of emerging technologies for hearing loss. [Work supported by NIDCD R01 DC 13267.]

4aPP44. Factors influencing auditory localization with deep insertion hearing aids or earplugs. Douglas Brungart (Walter Reed NMMC, 4401 Holly Ridge Rd., Rockville, MD 20853, dsbrungart@gmail.com), Nathaniel Spencer (AFRL/711th HPW, Wright-Patterson AFB, OH), Nina Pryor (AFRL/711th HPW, WPAFB, OH), Eric R. Thompson (AFRL/711th HPW, Wright-Patterson AFB, OH), Nandini Iyer (AFRL/711th HPW, Wpafb, OH), Griffin D. Romigh, and Brian Simpson (AFRL/711th HPW, Wright-Patterson AFB, OH)

Virtual localization experiments have demonstrated that Head Related Transfer Functions measured a few millimeters inside a blocked ear canal can produce localization performance approaching what is measured in the free field. This suggests that an earplug inserted entirely inside the ear canal should be able to preserve normal localization performance so long as the stimulus is loud enough to overcome any insertion loss in the device at all frequencies. In this study, localization performance of normal-hearing listeners was measured with the Lyric extended wear hearing aid, both in active mode (where it acted like an electronic pass-through earplug) and in passive mode (where it acted like a passive hearing protector). In an active mode, localization accuracy approached the open-ear condition. However, under the passive condition, localization was much worse than with the open ear even at high stimulus levels where the full spectrum should have been audible. This result suggests there may be fundamental limitations on localization accuracy with passive hearing protection that are unrelated to the directionality of the HRTF. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of the Army/Air Force, Department of Defense, or U.S. Government.]

4aPP45. The effects of age on narrowband and broadband measures of spectral processing in listeners with hearing loss. Kristi Oeding and Evelyn E. Davies-Venn (SLHS, Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55445, venn@umn.edu)

The propensity for degraded auditory perception increases with age. Several studies have shown that while age-related hearing impairment explains a high percentage of the often-reported degradation in auditory perception, there still remain some effects that can only be attributed to the aging process. Even though some classic studies have shown that spectral processing may be immune to age-related degradation, some recent work with broadband measures of spectral processing appears to challenge this notion. This study evaluated the effect of age on narrowband and broadband spectral processing abilities for individuals with mild-to-moderate hearing loss. We controlled for the amount of hearing loss and measured auditory filter bandwidths using notched-noise masking and spectral modulation detection using rippled noise in the same cohort of listeners. Results to date suggest that broadband spectral processing, which uses stimuli that share ecological validity with speech, may be more sensitive to age-related changes in spectral processing compared to narrowband spectral processing.

4aPP46. Executive functions predict improvements in pure-tone thresholds for children with normal hearing and children with hearing loss. Ryan W. McCreery, Lori Leibold (Audiol., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, ryan.mccreery@boystown.org), Emily Buss (Univ. of North Carolina, Chapel Hill, NC), and Elizabeth Walker (Univ. of Iowa, Iowa City, IA)

Detection of pure tones is known to improve in early childhood. The ability to suppress self-generated noise during detection tasks has been identified as a contributing factor to higher audiometric thresholds in children. Data from children with hearing loss indicate that children with thresholds that are above the level of self-generated noise do not show the same improvements across age as peers who can hear their self-generated noise. In an effort to examine the specific cognitive mechanisms that affect this process, audiometric thresholds and executive functions skills were measured in a group of 213 1st and 3rd grade children (84 children with normal hearing, 129 children with hearing loss). The relationship between selective attention and vigilance and thresholds were assessed for both groups. Children with stronger selective attention and vigilance skills had lower pure-tone thresholds than peers with poorer skills in these domains. The previous finding that children only showed an improvement in thresholds when they could hear their self-generated noise was also replicated. Audiometric test equipment that can monitor the sound level in the ear canal may help to increase the accuracy of clinical pure-tone detection tasks in children.

4aPP47. Just-noticeable differences of fundamental frequency change in Mandarin-speaking children with cochlear implants. Wanting Huang, Lena Wong (Div. of Speech and Hearing Sci., The Univ. of Hong Kong, Rm. 730, Meng Wah Complex, Hong Kong 999077, Hong Kong, wtwong88@connect.hku.hk), and Fei Chen (Dept. of Elec. and Electron. Eng., Southern Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Fundamental frequency (F0) provides the primary acoustic cue for lexical tone perception in tonal languages but is poorly presented in cochlear implants (CIs). Currently, there is still a lack of understanding on the sensitivity to F0 information in CI users speaking tonal languages. In the present study, just-noticeable differences (JNDs) of F0 contour and F0 level change in Mandarin-speaking kindergarten-aged children with CIs were measured and compared with those in age-matched normal-hearing (NH) peers. Statistical analysis showed that both JND of F0 contour change (JND-C) and JND

of F0 level change (JND-L) were significantly larger in CI group than in NH group. Furthermore, within-group comparison of JND-C and JND-L found that JND-C was significantly smaller than JND-L among children with CIs; however, opposite pattern was observed among children with normal hearing. The contrary sensitivity to F0 contour and F0 level change between children with CIs and children with normal hearing suggest discrepant mechanisms of F0 processing in these two groups as a result of hearing experience.

4aPP48. Assessment of a feasible virtual acoustics method for testing hearing aids using the Hearing-Aid Speech Perception Index. Sungbeen Cho, Scott Aker, and Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd., Elborn College 1212, London, ON N6G 1H1, Canada, scho255@uwo.ca)

Significant differences have been found in hearing aid (HA) performance between laboratory and real world test environments. Virtual sound environments provide a degree of control and reproducibility which is lacking in real world testing but may require an impractical number of loudspeakers. We assessed the accuracy of a simulation approach in which sources' direct sound is delivered by single loudspeakers while room acoustics are reproduced using low-order Ambisonics and a small number of loudspeakers. In a large office, we recorded binaural hearing aid output in response to sentence targets and babble noise presented at various levels and from various combinations of four loudspeakers surrounding a manikin. We measured the loudspeakers' room impulse responses (IRs) using a 32-channel spherical microphone array (Eigenmike), and split the IRs into "direct sound" and "room sound" portions. In an anechoic chamber, the original acoustics were simulated using Ambisonics or discrete loudspeakers for each source's direct portion and Ambisonics for the room portion. Ambisonic order and/or number of playback loudspeakers were also varied. HA output in the simulations was recorded using the manikin and assessed by comparing Hearing-Aid Speech Perception Index (HASPI) values computed on the simulation recordings with those made in the original room.

THURSDAY MORNING, 16 MAY 2019 STOPHER, 8:00 A.M. TO 11:00 A.M.

Session 4aSA

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

Benjamin Shafer, Cochair
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Contributed Papers

8:00

4aSA1. Power transmission metrics and applications in the design of quiet structures. Jonathan D. Young and Kyle R. Myers (Appl. Res. Lab., Penn State Univ., 3220B G Thomas Water Tunl, P.O. Box 30, State College, PA 16804, krm25@arl.psu.edu)

Connected structures subject to applied dynamic loads transfer vibrational energy through their connecting junctions. Identifying the dominant paths of transmission and characterizing the power flow through those paths is important for designing a quiet structure. When the connection type is known, one way of characterizing the transmitted power flow is to identify which degrees of freedom (i.e., translational and rotational) is most responsible for transmission through the junction. Another way could identify vibrational modes that dominate the transmission. This research presents several examples that characterize power flow between structures in physical and modal space. The structures examined here may be connected through springs, point impedances, or a generalized impedance matrix. Key questions considered are how changes to the system affect transmitted power, and how the results can be used to design quieter structures.

8:15

4aSA2. Development of vibrational metrics for internal damage scenarios of a scaled Transnuclear-32 dry storage cask for spent nuclear fuel. Kevin Y. Lin (Phys. and Astronomy, and National Ctr. for Physical Acoust., Univ. of Mississippi, 145 Hill Dr., Oxford, MS 38677-1848, klin@go.olemiss.edu), Joel Mobley, Wayne E. Prather, Zhiqu Lu, Gautam Priyadarshan, and Josh R. Gladden (Phys. and Astronomy, and National Ctr. for Physical Acoust., Univ. of Mississippi, University, MS)

The assessment of the internal structural integrity of dry storage casks for used high burnup nuclear fuel assemblies is of critical importance before transporting these to permanent repositories. The large size and structural complexity of the Transnuclear-32 (TN-32) cask as well as the inability to access its interior make this a challenging task. To address these difficulties, we use an active acoustics approach to develop metrics that are sensitive to the internal configuration of these casks. A 6:1 scaled model of the TN-32 cask was constructed in order to study the internal configuration of the fuel assemblies including various damage scenarios. Each mock-up fuel assembly consists of bundled steel rods, and their structural failure is mimicked

by steel shot of equal weight. This talk will report the amplitude- and phase-based active acoustics metrics that we developed to characterize different levels of internal damage. Our studies indicate that vibrometric signatures of various internal conditions can be measured using sources and sensors mounted on the exterior shell. Our current methodology is sensitive enough to detect structural failures at the single fuel assembly level. [Work supported by DOE NEUP Award No. DENE0008400.]

8:30

4aSA3. Working gases based scaling-down limitations for thermoacoustic coolers: Miniaturization approach. Anas M. Abdelrahman and Xiaoqing Zhang (Dept. of Refrigeration and Cryogenics, School of Energy and Power Eng., Huazhong Univ. of Sci. and Technolgy, 1037#, Luoyu Rd., Hong Shan District, Wuhan 430074, China, arahman@hust.edu.cn)

Regarding miniaturization of thermoacoustic coolers for thermal management purposes, working gases play a key role as the primary media responsible for producing the so-called "thermoacoustic effect" with their interaction with solid media (i.e., secondary media) within stacks or regenerators. However, the role of working gases in limiting scaling-down of thermoacoustic coolers still needs more investigations compared to addressed operational parameters (i.e., mean pressure, temperature difference across stack, etc.). In the present study, a theoretical computational analysis, based on published literature work, would be conducted to investigate allowable minimum sizes of standing-wave thermoacoustic coolers under the effects of working gases thermo-physical properties with considering adverse effects limitation of thermal conduction losses. Different working gases including air and either pure or mixture noble gases have been used for such geometrical scaling-down analysis under specific operating conditions. Moreover, cooling power was focused here as the more desirable performance indicator rather than the efficiency. The results had revealed the cooling capability at different scale levels based on different working gases, which make gases properties significantly contribute to scalability of thermoacoustic coolers to meet the cooling needs for micro-electronics. In addition, more research work will be devoted to other scaling-down issues of thermoacoustic systems.

8:45

4aSA4. Suspension optimization for a compressor assembled on a refrigerator. Alexandre A. Pescador Sarda (Mech. Eng., UFPR, Av. Cel. Francisco H. dos Santos, 100, Curitiba, PR 81530000, Brazil, pescador@ufpr.br) and Arcanjo Lenzi (Engenharia Mecânica, UFSC, Florianópolis, SC, Brazil)

Noise annoyance generated by electrical machines is evaluated based on the sound power level (SPL) parameter measured in a reverberationroom or using a semi-anechoic chamber. However, when the machine is assembled in a base plate, the SPL can be altered depending on the new system configuration and the way the machine is assembled in the final product. The vibration generated in the electrical machine is transmitted to the base resulting in noise at the surface. The aim of this study was to model the suspension of a general machine assembled in a flexible base and minimize the vibratory power flow transmitted to the base through an optimization process, taking into account parameters such as the spring inclination, stiffness, and suspension damping. Decreasing the power flow to the base results in a reduction in the global levels of noise and vibration at the base plate.

9:00

4aSA5. Broadband sound localization with gradient helical structure. Jie Zhu (Mech. Eng., Hong Kong Polytechnic Univ., FG603, Kowloon, Hong Kong, jiezhu@polyu.edu.hk)

Acoustic sensors or microphones are essential equipments for the detection of sound signals. However, sound signal suffers from the inevitable attenuation due to many reasons, such as diffusion, damping, thermal, and viscous loss. To solve such a problem, we introduce a gradient acoustic metamaterial to magnify the sound signal before it can be coverted to the electrical signal by introducing a gradient refractive index along the sound propagating route. In this case, the acoustic signal can be magnified as the wave is compressed by the gradient increasing index. The helicoid metamaterial has the advantage to adjust acoustic parameters flexibility and continuity by changing the pitch of the blades. This design is of great significance to improve the working condition of signal detection and may contribute to the design of other sensors.

9:15

4aSA6. Resonance frequencies of a spherical aluminum shell subject to prestress from internal fluid pressure. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu) and Natalie Harris (Phys., Whitman College, Walla Walla, WA)

Vibration measurements of a spherical aluminum shell (6 in. diameter) filled with water show that the resonance frequencies of the shell shift higher or lower with increasing water pressure, depending on the specific mode of vibration. For a given mode, the rate of frequency shift with pressure change $\Delta f/\Delta p$ is approximately linear for gauge pressures up to 100 psi. Frequency shifts were detected for pressure changes as small as 0.2 psi or 10 mm Hg. Observations of positive frequency shifts are consistent with previous studies (from the 1950s) involving submerged cylindrical shells subject to much larger pressures. Analysis from this era suggests that the phenomenon is due to geometric nonlinearity; however, the negative frequency shift observed with low order modes is not predicted by this theory. The feasibility of developing a noninvasive method for monitoring intracranial pressure using shifts in skull resonance frequencies will also be discussed.

9:30

4aSA7. On the investigation of the natural mode characteristics of an internal supporting substructure interacting with a submerged main structures in terms of acoustic radiations. Pei-Tai Chen (Dept. of System Eng. and Naval Architecture, National Taiwan Ocean Univ., No. 2,Pei-Ning Rd. Keelung 20224, Taiwan, ptchen@mail.ntou.edu.tw)

This paper investigates acoustic radiation characteristics of an internal structure supported in a submerged main structure. An exciting force is applied on the internal structure where vibration is transmitted across the interfacial boundary between these two structures and radiating acoustic power into water through the surface contacting with water. In the previous study (Chen, in *176th ASA meeting*, Vol. 144, p. 1680), compliant matrices describing the internal structure and the main submerged structure are proposed, which are both symmetric matrices where the imaginary part of the compliance matrix of the submerged main structure is responsible for acoustic radiation, whereas the matrix for the internal supporting structure is real. These two compliance matrices which are defined on the interfacial boundary fully describing the submerged structural dynamics of this coupled system. It was shown that the parameters, such as thickness and stiffened plate, of the internal structures are very sensitive variations of acoustic radiations, although the stiffness of the internal structures are much lower as compared with the main submerged structure. The present study addresses the natural mode characteristics of the internal structure, such as natural frequencies and mode shapes, affecting power flow across the connecting boundary and thus radiating into water. Two sets of modal expansions for forces and displacements defined on the connecting interfacial boundary are established to investigate acoustic radiations.

9:45–10:00 Break

10:00

4aSA8. A parametric resonance based capacitive ultrasonic transducer for wireless power transfer in air. Sushruta Surappa and F. Levent Degertekin (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Rm. 318, Atlanta 30332, GA, sushsurappa@gatech.edu)

Capacitive ultrasonic transducers have been in use for many years for various applications such as medical imaging, wireless power, sensing, and nondestructive testing. Typically, capacitive transducers require a DC bias or electret in order to operate efficiently and with high sensitivity. This makes them less desirable for applications such as wireless power transfer or energy harvesting, where a passive piezoelectric transducer may be preferred. Recently, it was shown that the requirement of a DC bias can be overcome by driving the capacitive transducer into parametric resonance

using ultrasound (Surappa *et al*., *Appl. Phys. Lett*. **111**(4), 043503). In this work, we present the operation of the first ever biasless, chargeless capacitive parametric ultrasonic transducer (CPUT) in air and demonstrate the utility of such a system for acoustic power transfer and sensing. Experiments performed in air show that the CPUT has an open-circuit sensitivity of more than 100 mV/Pa and is able to recover 31 μW at a distance of 10 cm from a 50 kHz ultrasonic source in the absence of a DC bias.

10:15

4aSA9. Micro-electro-mechanical system multi-resonant accelerometer for auditory prosthetics. Alison Hake and Karl Grosh (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, aehake@umich.edu)

Low usage of auditory prosthetics such as hearing aids can be attributed to several factors including appearance, stigma, acoustic feedback, and poor performance in noisy environments. Cochlear implants suffer from similar stigma; furthermore, external components of the system must be removed for sleeping, bathing, and physical exercise. Designing a completely implantable system that includes a sensor placed in the middle ear can mitigate these issues. Our approach is to use a miniature piezoelectric accelerometer to sense the vibration of the middle ear ossicles rather than employing an external or subcutaneous microphone to sense incoming sound. Results of our previous work showed the potential of this approach using a traditional single-resonance sensor. We seek to improve the low-frequency input referred noise (IRN) of the system by using a new architecture consisting of an array of piezoelectric MEMS beams with different resonant frequencies. The beams are connected electrically in a manner that increases the system sensitivity over the bandwidth of interest, thereby decreasing the IRN. Preliminary analytic studies have illustrated that 10 parallel-connected beams can improve the IRN by approximately 45 dB at 100 Hz. This method could further miniaturize sensors capable of detecting ossicular vibration from 100 Hz to 8 kHz.

10:30

4aSA10. Method for low frequency sound reflection coefficient measurements in a compact space. Xiuyuan Peng, Junfei Li, Chen Shen, Kiegan Lenihan, and Steven Cummer (Elec. and Comput. Eng., Duke Univ., 101 Sci. Dr., Durham, NC 27705, xp21@duke.edu)

Here, we present a method to measure the quasi-normal reflection coefficients of sound absorbing materials in a compact space. A short incident sound pulse (length $<$ 3ms) is generated by a deconvolution method with the source speaker. Then, a stage-mounted microphone moves across the material surface and records the total (incident + scattered) sound field. By comparing the sound field with and without the presence of the sound absorbing material, the frequency-dependent reflection coefficients can be derived by extracting the corresponding frequency components from the sound pulse. Using this method, we can calculate the reflection coefficients of a 2 ft. by2 ft acoustic panel from 300 Hz to 2500 Hz within a 2 m by2 m by3 m lab space without anechoic coatings. Moreover, this method enables us to investigate the spatial inhomogeneity of the sound absorbing material by studying the amplitude/phase variation of reflection coefficients across the material surface. Compared with conventional measurement techniques for reflection/absorption coefficients, our method has the advantages of low cost, minimal requirements for the measurement environment and the ability to measure the reflection coefficients at different locations. The proposed method can be favorable for measuring reflection coefficients of two-dimensional acoustic panels/metamaterials at low frequencies.

10:45

4aSA11. Sound quality evaluation of automobile interior noise under transient and steady-state running conditions. Zhengwei Yang, Huihua Feng (Mech. Eng., Beijing Inst. of Technol., 5 South Zhongguancun St., Haidian District, Beijing 100081, China, yangzhengwei1991@126.com), and Shouwei Lu (Tech. Ctr., Baic Motor Powertrain Co., Ltd., Beijing, China)

Sound quality can reflect people's subjective auditory feelings; thus, it plays an important role in automobile interior noise evaluation in recent years. Most research focuses on steady-state running conditions. In this paper, automobile vibration and noise transfer paths were measured with the binaural transfer path analysis (BTPA) method under both transient and steady-state running conditions. Then, loudness, sharpness, roughness, and A-weighted sound pressure level were used for studying properties and differences of automobile interior noise among different running conditions. Moreover, an experiment was carried out for the subjects to mark the annoyance of all noise samples. After that, the artificial neural network was applied to create the sound quality model to assess automobile interior noise without subjective experiments. According to the scores and binaural transfer path synthesis(BTPS) results, structural improvement methods were proposed for better sound quality of the automobile.

Session 4aSP

Signal Processing in Acoustics: Emerging Techniques for Acoustic Signal Processing

Michael J. Roan, Chair
Mechanical Engineering, Virginia Tech, 111 Randolph Hall, 460 Old Turner St., Blacksburg, VA 24061

Contributed Papers

8:30

4aSP1. Non-stationary source localization using direction of arrival measurements from multiple in-air vector sensors. Miles B. Penhale and Andrew R. Barnard (Mech. Eng.-Eng. MecûÂnica, Michigan Technol. Univ., 1400 Townsend Dr., R.L. Smith MEEM Bldg., Houghton, MI 49931, mbpenhal@mtu.edu)

Vector sensors utilize a combination of pressure sensors, particle velocity sensors, or both, to determine the acoustic intensity magnitude and direction pointing toward an acoustic source. This acoustic intensity vector is often referred to as the Direction of Arrival (DOA). By combining DOA information from multiple vector sensor measurement locations, a sound source may be instantaneously localized. The majority of vector sensor research has been conducted for underwater applications. A few studies of in-air vector sensors, which utilize multiple microphones, have been conducted; however, the majority of them study stationary sound sources in a laboratory environment or non-real-world settings. The focus of this paper is to study in-air vector sensor capabilities when sensing non-stationary mechanical noise sources—specifically ground vehicles—in a non-laboratory environment where ambient noise may be present. The DOA measurements at multiple vector sensor locations are used to test the acoustic source localization potential for this method.

8:45

4aSP2. Audio and ultrasonic frequency response measurements using regularized inverse filtering. Yegor Sinelnikov (Acoust., Zebra, 126 Liberty Ave., Port Jefferson, NY 11777, ysinelnikov@yahoo.com), Andrew Boyden, Richard Lavery, and Russell Calvarese (Acoust., Zebra, Holtsville, NY)

Skyrocketing growth in digital voice communication technology prompts inclusion of ultrasonic microphones in modern smartphones. Inaudible ultrasonic waves provide safe and effective way for ambient intelligence in indoor environments. It is important to measure microphone response over combined audio and ultrasonic range. Conservative measurements based on frequency sweep present signal equalization challenge. Most ultrasonic tweeters produce narrowband ultrasonic tones, while audio speakers resonate below 1 kHz. We present a simple methodology based on emission of specially constructed noise stimulus that overcomes this limitation. The Ole Wolf 25 x16 mm 1 W 8 Ω speaker connected to B&K power amplifier delivered the stimulus. High frequency B&K type 2670 reference microphone was used to record responses at 192 kHz. Experiments were performed in two stages. First, the blue noise was emitted, its response was recorded, and new stimulus with flat spectra between 15–25 kHz was derived using regularized inverse filtering technique. Second, constructed stimulus was emitted and recorded by reference microphone and device under test. The frequency response was derived and found in good agreement with conventional stepped sweep measurements.

9:00

4aSP3. Classification of inter-floor noise type/position via supervised learning. Hwiyong Choi, Haesang Yang, Seungjun Lee, and Woojae Seong (Naval Architecture and Ocean Eng., Seoul National Univ., 1, Gwanak-ro, Bldg. 34, Rm. 305, Gwanak-gu 08826, Seoul, South Korea, its_me_chy@snu.ac.kr)

This work presents noise type/position classification of various inter-floor noises generated in a building which is a serious conflict issue in apartment complexes. For this study, a collection of inter-floor noise dataset is recorded with a single microphone. Noise types/positions are selected based on a report by the Floor management Center under Korea Environmental Corporation. Using a convolutional neural networks based classifier, the inter-floor noise signals converted to log-scaled Mel-spectrograms are classified into noise types or positions. Also, our model is evaluated on a standard environmental sound dataset ESC-50 to show extensibility on environmental sound classification.

9:15

4aSP4. Ultrasonic communications for real-time video-rate data transmission through tissue. Gizem Tabak, Michael L. Oelze (Univ. of Illinois at Urbana-Champaign, 1308 W Main St., 119 Coordinated Sci. Lab., Urbana, IL 61801, tabak2@illinois.edu), and Andrew Singer (Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Wireless implanted medical devices (IMDs), which communicate data wirelessly from sensors within the body to a receiver outside of the body, are poised to be significant contributors to medical diagnostics and treatment procedures. Currently, radio frequency (RF) electromagnetic waves are the most frequently used communication method for wireless IMDs. However, high attenuation of RF in the body and strict regulations on the RF frequency spectrum limit the data rates to 267 kbps for *in vivo* applications. Considering standard definition video streaming requires 1.2 Mbps, and HD requires greater than 3 Mbps, it is not possible to use RF communication, for example, in applications that require real-time video transmission and, possibly, intervention such as real-time video capsule endoscopy. In our work, we use ultrasonic waves to communicate through tissue at video-capable data rates (>1.2 Mbps). Previously, we demonstrated a 4 Mbps data rate with BER less than 1 10^{-4}through beef liver using small, 2-mm biocompatible transducers at 1.3 MHz. In this study, we will investigate the effects of target motion on data rates and demonstrate real-time communication links *in situ* in a dead animal and *in vivo*.

9:30

4aSP5. Personalizing head-related transfer functions using anthropometric measurements by combining two machine-learning models. MingYang Lee, Martin S. Lawless, and Melody Baglione (Mech. Eng., The Cooper Union for the Advancement of Sci. and Art, 41 Cooper Sq., New York, NY 10003, lee11@cooper.edu)

Virtual reality (VR) requires rendering accurate head-related transfer functions (HRTF) to ensure a realistic and immersive virtual auditory space.

An HRTF characterizes how each ear receives sound from a certain location in space based on the shape of the head, torso, and pinnae, and provides a unique head-related impulse response (HRIR) for each given source location. Since HRTFs are person-specific and difficult to measure, recent research has utilized pre-existing HRTF databases and anthropometric measurements to generate personalized HRTFs with machine learning algorithms. This study investigates a personalization method that estimates the shape of each ear's HRIR and interaural time differences (ITD) between the two ears in separate models. In the proposed method, the shape of the HRIR is estimated with an artificial neural network (ANN) trained with time-aligned HRIRs from the CIPIC database, eliminating between-subject timing differences. A regression tree is used to estimate the ITDs, which are integer sample delays between the left and right ears. A localization test with a VR headset was conducted to evaluate the perceptual accuracy of the personalized HRTFs. Subjects completed the test with both a pre-selected average HRTF and their personalized HRTF to compare localization errors between the two conditions.

9:45–10:00 Break

10:00

4aSP6. Improving autonomous vehicle safety—The use of convolutional neural networks for the detection of warning sounds. Ethan Wagner and Eoin A. King (Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu)

In cities across the world everyday, people use and process acoustic alerts to safely interact in and amongst traffic. With the advent of autonomous vehicles (AVs), the manner in which these new vehicles can use these acoustic cues to supplement their decision making process is unclear. This will be especially important during the prolonged period of mixed vehicles sharing the road. One solution may lie in the advancement of machine learning techniques; it has become possible to "teach" a machine (or a vehicle) to recognize certain sounds. This paper reports on an ongoing project with the objective of identifying emergency vehicles sirens in traffic and alerting the vehicle to take rapid evasive action. In particular, we report on the use of a deep layer Convolutional Neural Network (CNN) trained to recognize emergency sirens. We retrained a CNN (AlexNet) to recognize sirens in real time. To utilize this network, samples from the ESC-50 dataset for environmental sound classification were processed and each converted to a spectrogram. This CNN can be used in conjunction with a microphone array to accurately recognize sirens in traffic and identify the direction from which the emergency vehicle is approaching.

10:15

4aSP7. A review of techniques for ultrasonic indoor localization systems. Joaquín Aparicio (Dept. of Informatics, Univ. of Oslo, Gaustadalléen 23B, Ole-Johan Dahls hus, Oslo 0851, Norway, joaqapar@ifi.uio.no), Fernando J. Álvarez (Sensory Systems Res. Group, Univ. of Extremadura, Badajoz, Spain), Álvaro Hernández (Electronics Dept., Univ. of Alcalá, Alcalá de Henares, Spain), and Sverre Holm (Dept. of Informatics, Univ. of Oslo, Oslo, Norway)

Accurate localization in indoor environments is crucial for the correct operation of location-aware and augmented reality applications, indoor navigation, and inventory management, among others. Magnetic, radiofrequency and inertial navigation systems typically provide room or meter-level accuracy. Despite being the most widely used, they are affected by error drifts or changes in the environment where they operate, and their accuracy is a drawback for certain applications, such as navigation. Optical-based systems provide better accuracy, but they can be expensive, and they are not privacy-oriented. Ultrasonic positioning systems can also give room-level accuracy, as acoustic propagation is contained within the room walls, helping resolve room or floor-level ambiguities of radio systems. They can even achieve centimeter-level accuracy and ensure privacy, while being low cost. These properties highlight acoustics as a versatile technology for different indoor localization applications, as stated by the research published almost over the last three decades. In this work, the operating principles of the different techniques employed by acoustic positioning systems are reviewed, covering narrowband and wideband systems (including the differences between coded and uncoded transmissions), fingerprinting, and the most recent systems based on machine learning.

10:30

4aSP8. Extending bandwidth for sound power measurements. Michael C. Mortenson, Suzanna Gilbert, Tracianne B. Neilsen, Kent L. Gee, and Scott D. Sommerfeldt (Brigham Young Univ., N 283 ESC, Provo, UT 84602, michaelcmort@byu.edu)

Sound power is often measured using the intensity-based engineering standard ANSI S12.12-1992. Traditional methods for intensity-based sound power estimation are limited in bandwidth at low frequencies by phase mismatch between microphones and at high frequencies by microphone spacing—with errors occurring well below the spatial Nyquist frequency. The PAGE (Phase and Amplitude Gradient Estimation) method has been used to extend the bandwidth of intensity calculations [Gee *et al.*, *J. Acoust. Soc. Am.* **141**(4), EL357–EL362 (2017)]. This paper examines the efficacy of the PAGE method to overcome bandwidth limitations in estimating sound power. Specifically, the sound fields from three sources—a blender, a vacuum cleaner, and a dodecahedron speaker—were measured according to ANSI S12.12-1992. The sound power was computed for each source using both the traditional and PAGE methods. The resulting intensity-based sound power estimates are compared against sound power measurements obtained according to the scientific-grade ISO 3741:1999 standard. The PAGE method increases the bandwidth over which reliable estimates are achievable for intensity-based sound power estimates, even exceeding the spatial Nyquist frequency when phase unwrapping is successful. Thus, using existing equipment, industry professionals can extend the bandwidth of sound power estimates with the PAGE method. [Work supported by NSF.]

10:45

4aSP9. On the viability of the complex-intensity-based near-field acoustical holography method. Caleb B. Goates, Scott D. Sommerfeldt, Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com)

Because of the instrumentation difficulties of Fourier acoustical holography, it is desirable to find a holography method that does not require reference signals during array scanning. Among the methods that have been investigated to remove the need for references are those based on acoustic intensity measurements, including the complex-intensity-based near-field acoustical holography (CIBNAH) method [A. Nejade, *J. Sound Vib.* **333**(16), 3598–3608 (2014); *Appl. Acoust.* **116**, 348–356 (2017)]. The CIBNAH method has previously been applied to simple contrived sources and real-world machinery but has not been verified using analytical source models. This work shows the application of CIBNAH to an analytical model of a simply-supported plate, revealing key shortcomings of the method. The theory behind CIBNAH is discussed in light of these shortcomings. It is shown that while CIBNAH may be useful for finding radiation hot spots, it is not an adequate method to overcome the need for references in scan-based acoustical holography. [Work supported by NSF.]

11:00

4aSP10. Employing braid theory to disentangle overlapping oceanic events in the shallow water acoustic channel. Ananya Sen Gupta and Ryan McCarthy (Elec. and Comput. Eng., Univ. of Iowa, 4016 Seamans Ctr. for the Eng. Arts and Sci., Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

The shallow water acoustic channel is well-known to exhibit rapid temporal fluctuations due to multipath effects, fluid motion as well as focusing events in the ocean. These events often overlap temporally thus contributing in unpredictable ways to the channel delay spread. We will present recent results from employing geometric signal processing techniques to shallow water channel estimation and will introduce new ways to identify and disambiguate overlapping oceanic phenomena within the channel delay spread. Specifically, we will explore how geometric braid theory and related operations can be employed to disentangle overlapping channel braids, and explore the efficacy of such signal processing techniques across diverse channel representations. Finally, we will discuss the importance of capturing oceanic phenomena at the appropriate resolution of time, frequency, and sparsity and the trade-offs in accuracy involved based on the chosen channel representation(s).

Session 4aUW

Underwater Acoustics, Physical Acoustics, Structural Acoustics and Vibration, Acoustical Oceanography, Signal Processing in Acoustics, and Computational Acoustics: Uncertainty in Propagation Prediction

Jennifer Cooper, Cochair
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D. Keith Wilson, Cochair
US Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03755-1290

Invited Papers

8:20

4aUW1. Uncertainty quantification for right-sizing computational models of sound propagation in the atmospheric boundary layer. Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., 590 Holloway Rd., Annapolis, MD 21402, pettitcl@usna.edu), D. K. Wilson, and Carl R. Hart (U.S. Army Cold Regions Res. and Eng. Lab., Hanover, NH)

Comprehensive modeling of sound propagation through the atmospheric boundary layer is viewed as a judicious combination of accurate computational mechanics models and uncertainty quantification (UQ) methods. The role of numerical models is to represent nominally deterministic phenomena, e.g., geometrical spreading, ground interactions, refraction by mean gradients of wind and temperature. The role of UQ is to characterize the consequences of fundamentally non-deterministic and imprecisely known factors that affect propagation, e.g., turbulence in the atmospheric boundary layer, complex terrain features, and overly sparse spatio-temporal sampling of propagation parameters. High-fidelity wave propagation mechanics cannot compensate for inherent randomness in the environment and insufficient data on the parameters. When uncertainty is significant, the computational cost of high-fidelity models might be better invested in more ensemble simulations with medium-fidelity models and quantifying the payoff from more data about the environment. Work in recent years along three thrusts to enable this form of comprehensive modeling is reviewed: (1) Surrogate modeling based on cluster-weighted models, which are a type of probabilistic generative model, and on statistical learning methods, (2) global sensitivity analysis for assessing the importance of model parameters, and (3) a computational mechanics error budget for rationally analyzing the importance of various sources of uncertainty.

8:40

4aUW2. Statistics of acoustic waves propagating through turbulent media. Philippe Blanc-Benon (Ecole Centrale de Lyon, Université de Lyon, LMFA UMR CNRS 5509, 36 Ave. Guy de Collongue, Ecully 69134 Cedex, France, Philippe.blanc-benon@ec-lyon.fr)

Propagation of acoustic waves through atmospheric turbulence is relevant to different problems : outdoor sound propagation, blast waves generated from explosions or gunshots, propagation of sonic booms. While propagating in turbulent air, acoustic waves are distorted by the combined effects of diffraction and scattering induced by atmospheric inhomogeneities. Accurately controlled experiments are needed to validate theoretical models for sound propagation in inhomogeneous media. In this paper, probability distribution functions will be presented for linear and nonlinear acoustic wave propagation through thermal or kinematic turbulence. Experimental data will be compared with numerical simulation using parabolic approaches. [Work supported by the Labex CeLyA of Université de Lyon, operated by the French National Research Agency (ANR-10-LABX-0060/ ANR-11- IDEX-0007).]

Contributed Paper

9:00

4aUW3. Effect of uncertainty in meteorological conditions on aircraft noise levels. Harshal Prakash Patankar and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, hpp103@psu.edu)

To comply with noise regulations and to plan infrastructure around airports, there is a need to accurately predict aircraft noise levels. Even when high fidelity noise propagation models are used, the accuracy of noise level predictions can be affected by uncertainty in the input parameters. This work looks at the change in SPLs due to these uncertainties. To incorporate the effect of meteorological uncertainties, the methodology presented in Wilson *et al.* [*JASA*(2014)] is extended to the geometry of aircraft noise propagation. To allow for faster computations, a simplified version of the NORD 2000 noise propagation method is used. This work explores the effect of individual meteorological uncertainties in the propagation path (such as temperature profile and wind profile) as well as their combined effect on the SPLs for terminal area aircraft altitudes. [Work supported by the U.S. Federal Aviation Administration Office of Environment and Energy through ASCENT, the FAA Center of Excellence for Alternative Jet Fuels and the Environment, project 40 through FAA Award No. 13-C-AJFE-PSU under the supervision of Hua He. Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

9:15

4aUW4. Methods for including uncertainty in modeling and simulation of acoustic propagation in ocean waveguides. Steven I. Finette (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Two main classes of uncertainty are associated with modeling and simulation of acoustic fields in ocean waveguides. These classes, generally known as aleatory and epistemic uncertainties, represent opposite ends of a spectrum of possible types of imperfect knowledge concerning the system. Aleatory uncertainty can be interpreted physically as natural ocean variability which is typically characterized by probability density functions, provided sufficient information is available to justify their specification. On the other hand, epistemic uncertainty is associated with incomplete scientific knowledge concerning some aspect(s) of the system under analysis. While epistemic uncertainty can be reduced by the inclusion of additional information, e.g., data refinements, acquisition of additional data or more realistic modeling of the system, aleatory uncertainty cannot be eliminated or reduced because it is judged by the modeler to be inherent in the structure of the system. Both types of uncertainty are discussed, with an emphasis on the aleatory contribution based on stochastic basis expansions. Tradeoffs with Monte-Carlo and Bayesian methods are also considered. When density functions are not available, weaker inferential estimates based on epistemic approaches are appropriate and a hybrid aleatory-epistemic framework is outlined for treating these situations. [Work supported by the Office of Naval Research.]

9:35

4aUW5. Strategies for linking ocean state uncertainty to acoustic uncertainty. Timothy F. Duda, Weifeng G. Zhang, Ying-Tsong Lin, and Arthur Newhall (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Estimating uncertainty of underwater sound fields caused by partially known environmental conditions is a broad topic with many branches because acoustic fields have many characteristics and many descriptors. Specific field characteristics, each uncertain, influence sound use in specific ways. Multiple descriptors (parameters) of signals of interest, and noise, need to be adequately known to examine use scenarios. This is also true for field simulation, processor simulation, field uncertainty simulation, and processor uncertainty simulation. Parameters should be prioritized for efficient quantification of uncertainty. For example, spatial coherence uncertainty applies to array processing but possibly not to single-sensor processing. Linking environmental uncertainty to field uncertainty, then to task performance uncertainty suffers from the many degrees of freedom present in the environment, and the interconnected effects of the many variable environmental parameters. Here, linkage frameworks for sound within internal waves propagating in variable conditions are examined. Methods appropriate for deterministically defined wave groups are explored, as well as statistically described wave fields. First, wave parameters and parameter uncertainty are specified, then effects on the sound parameters are estimated, as well as derived quantities like probability of detection and direction of arrival. Canyon environments that we have studied with models provide one test bed.

9:55–10:10 Break

10:10

4aUW6. Bayesian calibration of underwater propagation models under uncertainty. Ralph T. Muehleisen, S H. Swift, and Qi Li (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg 221, Lemont, IL 60439, rmuehleisen@anl.gov)

Propagation models are notorious for the uncertainty of important parameters such as source strength, speed of sound profiles, and reflecting surface profiles. In many cases, one calibrates a model to measured data (e.g., sound levels or transmission loss) for the purposes of estimating these model parameters, i.e., for inverse modeling. Bayesian calibration methods have been developed that are extremely useful for calibration of models where parameters have high levels of uncertainty and problems may be under or over determined. The Kennedy and O'Hagen framework which uses a Gaussian process surrogate model to replace the model under calibration is especially useful when the underlying model is computationally expensive, and so, it may be difficult to apply many optimization based calibration methods. In this talk, we describe the application of the Kennedy and O'Hagen Bayesian Calibration framework to the calibration of an underwater ray tracing propagation model. The source strength and parameters for the sound speed profile are considered as highly uncertain. The Bayesian calibration technique is shown to improve model prediction and reduce the uncertainty of the unknown propagation parameters.

10:30

4aUW7. Quantifying ocean acoustic propagation uncertainty due to environmental uncertainty from ship-noise inversion. Stan E. Dosso (School of Earth & Ocean Sci, Univ of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca) and Dag Tollefsen (Defence Systems Div., Norwegian Defence Res. Establishment (FFI), Horten, Norway)

This paper considers the uncertainty in underwater acoustic propagation predictions that results from uncertainty in oceanic environmental parameters (water column and seabed) as estimated from acoustic inversion methods. The approach is general but is considered here for the inversion of acoustic fields from ships-of-opportunity, which represents a convenient and unobtrusive approach to environmental assessment, but with significant uncertainties for some environmental parameters. In this work, environmental uncertainties are quantified using trans-dimensional (trans-D) Bayesian inversion based on reversible-jump Markov-chain Monte Carlo sampling. Trans-D inversion considers the number of sediment layers in the seabed and/or the number of points defining the water-column sound-speed profile as unknown parameters, sampled probabilistically in the inversion. This approach numerically characterizes the posterior probability density of the environment using a large ensemble of dependent random samples of the model. Predicting acoustic propagation for a much-smaller, randomly-chosen subset of these model samples transforms environmental uncertainty to propagation uncertainty, which can be characterized spatially with standard statistical measures (standard deviation, credibility intervals, etc.). The approach is illustrated using simulations and noise from a large commercial ship recorded on an a horizontal array of hydrophones as part of the 2017 Seabed Characterization Experiment (SBCEX17) at the New England Mud Patch.

10:50

4aUW8. Variability of the modal response of a shallow water acoustic waveguide in the presence of uncertain sediment properties. Sheri Martinelli, Andrew S. Wixom (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, Mailstop 3230D, State College, PA 16804-0030, slm77@psu.edu), Mark Langhirt (Graduate Program in Acoust., The Penn State Univ., University Park, PA), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

Physical properties of the seabed comprise an important input set for physics-based modeling of sound propagation in littoral environments. Unfortunately this knowledge is often incomplete due to inherent space-time variability, yet deterministic models are still very much the standard for complex environments. Normal mode decomposition provides a well-established and understood framework for the study of underwater propagation. This work applies a generalized polynomial chaos expansion with stochastic collocation to propagate uncertain variables through a normal mode model, thus constructing expressions for the transmission loss (TL) and mode shapes themselves in terms of the input random variables. The goal of this work is to demonstrate the impact of imperfect knowledge of physical parameters on a deterministic computational model of the in-water acoustic field, and further study how well important information about model performance is captured by considering only first and second moments of the output distribution. An emphasis on material properties and geometry of sediment layering in the ocean bottom serves to isolate the effects of sediment uncertainty. Such a study can provide guidance for the use of deterministic models in performance prediction and has consequences for geo-acoustic inversion.

11:10

4aUW9. Three-dimensional underwater sound pressure sensitivity in oceanic 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)

Physical oceanographic processes, marine geological features, sub-bottom geoacoustic structure, sea surface disturbances, etc., can either individually or jointly affect underwater sound propagation in the ocean and cause significant temporal and spatial variability in the sound pressure field. The primary goal of this study is to develop a numerical scheme to determine the sound pressure sensitivity in response to variations of index of refraction due to changes of environmental conditions. This sensitivity analysis is an exention of the Born approximation which assumes perturbations at infinitesimal points. To handle disturbance within a finite volume, an improved sensitivity kernel is derived from a higher-order parabolic-equation (PE) approximation. With this sensitivity kernel, we can analyze the spatial distribution and the temporal evolution of the acoustic sensitivity field in complex oceanic environments. This paper will present numerical examples of three-dimensional (3D) sound propagation in continental slopes, submarine canyons, and nonlinear internal wave fields. Discussions on other applications, including uncertainty quantification of transmission loss prediction and adjoint models for 3D acoustic inversions, will also be provided. [Work supported by the Office of Naval Research.]

Contributed Papers

11:30

4aUW10. Machine learning methods for estimating probability density functions of transmission loss: Robustness to source frequency and depth. Brandon M. Lee and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, leebm@umich.edu)

Predicted values of transmission loss (TL) in ocean environments are sensitive to environmental uncertainties. The resulting predicted-TL uncertainty can be quantified via the probability density function (PDF) of TL. Monte Carlo methods can determine the PDF of TL but typically require thousands of field calculations, making them inappropriate for real-time applications. Thus, a variety of alternative techniques based on polynomial chaos, field shifting, modal propagation in ocean waveguides, and spatial variations of TL near the point(s) of interest have been proposed. Recently, an approach to estimating the PDF of TL based on nominal TL, ocean environmental parameters, and machine learning was found to have a success rate of 95% with constant source depth (91 m) and frequency (100 Hz) when tested on 657,775 receiver locations within 100 randomly selected ocean environments. This presentation describes an extension of this approach and its success predicting the PDF of TL for different source depths and frequencies for ranges up to 100 km. This increase in the size of the parameter space furthers the need for a sophisticated method of choosing training examples. Such a method is proposed, and its performance is compared to that of prior techniques. [Work supported by ONR.]

11:45

4aUW11. Model-data comparison of sound propagation in a glacierized fjord with a variable ice top-boundary layer. Matthew C. Zeh (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 204 East Dean Keeton, Stop C2200, Austin, TX 78712-1591, mzeh@utexas.edu), Oskar Glowacki (Marine Physical Lab, Scripps Inst. of Oceanogr., Warsaw, Poland), Grant B. Deane (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA), Megan S. Ballard (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Erin C. Pettit (College of Earth, Ocean, and Atmospheric Sci., Oregon State Univ., Fairbanks, AL), and Preston S. Wilson (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Transmission loss measurements were conducted in the meltwater-modified surface layer near Hansbreen Glacier in Hornsund Fjord in southwestern Svalbard in September 2017 [Deane and Glowacki, *JASA* **143**, 1711 (2018)]. An m-sequence source signal (149 dB re 1 μPa, 11 kHz carrier frequency) was tethered at 7 m depth to a boat drifting from 0 to 200 m. This signal was received by two Hitech HTI-96 hydrophones at 8 and 17 m depth deployed from a stationary boat anchored 500 m from the glacier. Within this environment, and typical for a glacierized fjord, regular calving events contributed to an ice mélange top boundary layer with larger icebergs occasionally obstructing the signal transmission path. The propagation environment was upward refracting, causing propagation sound to repeatedly reflect from the surface layer. A ray-based approach was applied to model the measured data. The variability of the top boundary was included in the model by incorporating surface scattering and inserting icebergs. Comparisons between several increasingly complex iterations of this model with the collected data will be presented. [Work supported by the NDSEG Fellowship, ONR Grant Nos. N00014-17-1-2633 and N00014-14-1-0213, and the Polish National Science Centre Grant No. 2013/11/N/ST10/01729.]

Session 4pAA

Architectural Acoustics and Signal Processing in Acoustics: Room Acoustics Modeling and Auralization

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Lauri Savioja, Cochair
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Invited Papers

1:15

4pAA1. Reverberation time and audibility in phased geometrical acoustics using plane or spherical wave reflection coefficients. Matthew Boucher (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681-2199, matthew.a.boucher@nasa.gov), Monika Rychtarikova (Faculty of Architecture, KU Leuven, Gent, Belgium), Lukas Zelem (Faculty of Civil Eng., Dept. of Architecture, STU Bratislava, Bratislava, Slovakia), Bert Pluymers, and Wim Desmet (Mech. Eng., Div. PMA, KU Leuven, Heverlee, Belgium)

In acoustical spaces, room acoustic parameters are often predicted using energy-based geometrical acoustics. For smaller rooms, interference among coherent reflections is taken into account by phased geometrical acoustics, which improves results for lower frequencies. The use of a spherical wave reflection coefficient improves the results further, yet the impact on room acoustic parameters is not fully known. This work focuses on the differences in predicted reverberation time when using plane or spherical wave reflection coefficients. The differences are analyzed for a variety of boundary conditions, including non-uniform distributions of absorption, in medium-sized rooms using a phased image source model. Since calculated differences are greater than the conventional just-noticeable-difference of 5% for reverberation time, a laboratory listening test is performed to confirm audibility of the modeled differences. Two narrow band noise stimuli (octave bands with central frequencies of 125 and 250 Hz) with a duration of 1 s were used for comparisons of 18 acoustic scenarios by means of a three-alternative forced choice method (3AFC). More than half of the listeners could hear the differences in all 36 cases. Statistically significant results (chi-squared test was used) were found in two thirds of the cases, corresponding to those with longer reverberation times.

1:35

4pAA2. Diffraction simulation from wedges to finite-sized plates based on the physical theory of diffraction. Ning Xiang, Anthony Savino (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu), and Aleksandra Rozynova (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Cambridge, MA)

Efficient predictions of sound diffraction around objects are of critical significance in room-acoustic simulations. An advanced diffraction theory has recently been investigated for potential applications in room acoustics for some semi-infinite, canonical wedges and for finite-sized rectangular plates [Rozynova and Xiang, *J. Acoust. Soc. Am.* **144** (to be published)]. The physical theory of diffraction (PTD) still relies on both geometrical and physical principles, yet emphasizes the physical one. Important features of the PTD approach are its computational efficiency and the high degree of accuracy for the diffracted sound field. This paper reviews the fringe field predictions of canonical semi-infinite wedges and further discusses solutions of diffraction problems on finite, rigid rectangular plates. The PTD is applied to approximate the solutions of a finite-sized, rigid rectangular plate that achieves high numerical efficiency. The PTD simulation allows sound diffraction contributions to be determined independently from two pairs of edges of the rigid plate, while ignoring the edge waves around the corner in far-field. This paper uses numerical implementations of the PTD predictions to demonstrate the simulation efficiency of the PTD in finite-sized objects. The numerical simulations are also validated by some preliminary experimental results carried out using an acoustic goniometer.

1:55

4pAA3. Individualization of head-related transfer functions using sparse representation approach. Zeng Xiangyang, Wang Lei, Lu Dongdong (Northwestern PolyTech. Univ., Xi'an, Shaanxi, 710070, Xi'an 710070, China, zenggxy@nwpu.edu.cn), and Huaizhen Cai (Dept. of Psych., Univ. at Buffalo, Buffalo, NY)

The individualization of Head-Related Transfer Functions (HRTFs) is an important issue for enhancing the performance of binaural auralization. In this paper, the HRTFs and anthropometric parameters of Chinese people were measured and analyzed. A sparse representation approach was suggested to synthesize individualized HRTFs with selected anthropometric features. The approach was compared with two other methods proposed in previous studies by comparing the spectrum distortion of each method for objective evaluation. Then, subjective experiments were conducted to investigate the performance of the optimized HRTFs in binaural localization. The evaluation results show that the proposed HRTFs individualization approach has smaller spectrum distortion and better localization performance than that of the reference methods.

2:15

4pAA4. Current trends in binaural auralization of microphone array recordings. Jens Ahrens, Carl Andersson (Chalmers Univ. of Technol., Sven Hultins Gata 8a, Gothenburg 412 58, Sweden, carl.andersson@chalmers.se), and Hannes Helmholz (Chalmers Univ. of Technol., Gotheburg, Sweden)

Many approaches for the capture and auralization of real acoustic spaces have been proposed over the past century. Limited spatial resolution on the capture side has typically been the factor that caused compromises in the achievable authenticity of the auralization. Recent advancements in the field of microphone arrays provide new perspective particularly for headphone-based auralization. It has been shown that head-tracked binaural auralization of the data captured by a bowling-ball-sized spherical array of around 90 microphones allows for creating signals at the ears of the listener that are perceptually almost indistinguishable from the ear signals that arise in the original space. Promising results have also been obtained based on smaller arrays with fewer microphones. In the present contribution, we provide an overview of the current activities in the research community and demonstrate the latest advancements and remaining challenges.

2:35

4pAA5. Simulation of a coupled room scenario based on geometrical acoustics simulation models. Lukas Aspöck and Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

As part of the most recent room acoustical simulation Round Robin, a coupled room scenario, consisting of a laboratory room and a reverberation chamber, is investigated. The evaluation of the participants' results, all using geometrical acoustics based simulation models, however, showed that the measured double slope of this scenario could not be matched by any of the six algorithms. In addition to the presentation of the measured and simulated results for this scenario, this work discusses the role the input data, especially the applied absorption coefficients, and the configuration of the simulation model. Eventually, additional results for one simulation model are presented to demonstrate the options and the limitations regarding the simulation of coupled volumes using geometrical acoustics models.

2:55–3:10 Break

3:10

4pAA6. Audibility of dispersion error in room acoustic finite-difference time-domain simulation in the presence of a single early reflection. Jukka Saarelma (Dept. of Comput. Sci., School of Sci., Aalto Univ., P.O. Box 13000, FI-00076 Aalto, Finland, jukka.saarelma@aalto.fi)

Finite-difference time-domain method has gained increasing interest for room acoustic prediction use. A well-known limitation of the method is a frequency and direction dependent dispersion error. In this study the audibility of dispersion error in the presence of a single surface reflection is measured. The threshold is measured for three different distance conditions with a fixed reflection arrival azimuth angle of 54.7 deg. The error is placed either in the direct path, or in the reflection path. Additionally, a qualitative follow-up experiment to evaluate how the measured thresholds reflect the audibility of error in short room responses is carried out. The results indicate that the threshold varies depending whether the error is in the direct path or in the reflection path. For transient signals, the threshold is higher when the error is located in the direct path, where as for speech signal the threshold is higher when it is located in the reflection path. Evidence is found that the error is detectable in rendered room responses at the measured threshold levels.

3:30

4pAA7. Benchmark analytical solutions for steady-state high frequency broadband sound fields in three rectangular enclosures. David Raudales (Physical Acoust. Branch Code 7130, Naval Res. Lab., 1420 W Abingdon Dr. Apt. 126, Alexandria, VA 22314, david.raudales@nrl.navy.mil), Donald B. Bliss (Mech. Eng., Duke Univ., Durham, NC), Krista Michalis (Mech. Eng., Duke Univ., Arlington, VA), Jerry W. Rouse (Mech. Eng., Duke Univ., Albuquerque, NM), and Linda P. Franzoni (Mech. Eng., Duke Univ., Durham, NC)

Analytical solutions are presented for interior broadband sound fields in three rectangular enclosures with absorption applied on the floor and ceiling, rigid sidewalls, and a vertically oriented dipole source. The solutions are intended to serve as benchmarks that can be used to assess the performance of broadband techniques, particularly energy-based methods, in a relatively straightforward configuration with precisely specified boundary conditions. A broadband Helmholtz solution is developed using a frequency-by-frequency modal approach to determine the exact band averaged mean-square pressures along spatial trajectories within each enclosure. Due to the specific choice of enclosure configuration and absorption distribution, an approximate specular solution can be obtained through a summation of uncorrelated image sources. Comparisons between the band averaged Helmholtz solution and the uncorrelated image solution reveal excellent agreement for a wide range of absorption levels and improve the understanding of correlation effects in broadband sound fields. A boundary element solution with diffuse boundaries is also presented which produces consistently higher mean-square pressures in comparison with the specular solution, emphasizing the careful attention that must be placed on correctly modeling reflecting boundaries and demonstrating the errors that can result from assuming a Lambertian surface.

3:50

4pAA8. Real-time auralization of a talker's own voice in virtual rooms. Jennifer Whiting, Timothy W. Leishman, and Traci Neilsen (Brigham Young Univ., 243 Eyring Sci. Cente, Provo, UT 84602, jkwhiting@outlook.com)

While much has been done in the field of sound auralization in virtual rooms, the problem of hearing one's own voice in these environments has received less attention. A robust and feasible system for real-time auralization of talkers who are also listeners is needed. To address this requirement, a real-time convolution system (RTCS) was designed with the specific goal of "placing" a talker/listener in virtual acoustic environments. This system necessitated the development of several tools and methods. Oral-binaural room impulse responses were measured and characterized for a variety of room. The RTCS improved on past systems, in part through the derivation and inclusion of compensation filters, which corrected the linear auditory distortions of the RTCS components. Objective measures in the time- and frequency-domains were developed to assess the validity of the system. A jury-based listening study also indicated that RTCS users could speak and listen to their own voices in the virtual acoustic environments in a natural manner.

4:10

4pAA9. Diffusion equation modeling in sound field analysis of disproportionate rooms and coupled spaces. Zühre Sü Gül (Architecture, Bilkent Univ., METU Technopolis KOSGEB-TEKMER No. 112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com)

This study discusses alternative models and methods to be applied in room acoustics estimations for specific room types including disproportionate rooms and rooms with coupled volumes. A recent prediction method, namely, diffusion equation model (DEM) in room acoustic applications is utilized in the methodology, and this method is compared to common models as of statistical theory, image-source or ray-tracing techniques. Both long enclosures as of subway stations and coupled volumes as of multi-domain monumental structures are special cases with specific interior sound fields. Statistical theory is not always reliable in such extraordinary room forms, while ray tracing may tend to over or under estimate certain acoustical parameters. Thus, the application of DEM in a finite element scheme for detailed sound energy decay and sound flow analysis are held over some case structures. The results are compared to field tested data and ray-tracing solutions. Pros and cons of DEM in comparison to different methods are searched in detail considering the efficiency in visualization, computational speed, and reliability of acoustical parameter results for specific room shapes.

4:30

4pAA10. Auralization of virtual concerts: A subjective evaluation comparing binaural and ambisonic rendering. David Thery (CHM, LIMSI-CNRS, Rue John Von Neumann, Orsay 75005, France, david.thery@limsi.fr)

Auralization renderings have reached a sufficient level of maturity that simulated auralizations can be comparable to measured ones. These auralizations can be rendered over a variety of sound systems, potentially combined with a visual model through VR interfaces. This study presents a perceptual evaluation of auralizations of a small ensemble virtual concert rendering, comparing a tracked binaural rendering to 2nd order Ambisonic rendering over a 32 loudspeaker array. The geometrical acoustic model of several actual performance spaces were created and then calibrated using *in situ* omni-directional room impulse response measurements. The performance stimuli consisted of 3 extracts of jazz anechoic recordings comprising trios and quartet ensembles, augmented by three-dimensional visual point-clouds of the musicians playing on stage. Participants of the listening test included a range of listening expertise level (acousticians, architects, students). Several room acoustical parameters were evaluated between rendering systems, seating positions, and rooms.

4:50

4pAA11. Discrete material optimization for wave-based acoustic design. Nicolas Morales (The Dept. of Comput. Sci., 201 S. Columbia St., Chapel Hill, NC 27599-3175, nmorales@cs.unc.edu) and Dinesh Manocha (Dept. of Comput. Sci., Univ. of Maryland, College Park, MD)

The problem of automatic design of acoustic spaces is prevalent in architecture and room acoustics. We present a novel algorithm to automatically compute the optimal materials of large architectural spaces. Our method uses discrete optimization techniques to determine the best material configuration for desired acoustic properties of a structure, while taking into account properties of real-world materials. An efficient acoustic wave solver is used to accurately compute the acoustic impulse responses that drive the optimization process. Our method is tested on various computer representations of real-world scenes where we show how new material characteristics can be computed to improve the scene's strength, clarity, and reverberation time.

5:10

4pAA12. Empirical evaluation of in-field, binaural record and playback reproduction. William Neale and Toby Terpstra (Visualization, Kineticorp, 6070 Greenwood Plaza Blvd., Ste. 200, Greenwood Village, CO 80111, wneale@kineticorp.com)

This research evaluates a methodology for calibrating in field, sounds for playback in a separate, interior environment. The ability to record sounds in the field and playback them accurately in a different environment is useful when the end user, or listener cannot be present at the location where the sound is being produced live. In forensics, for example, an expert or juror may need to evaluate an acoustic or auditory issue but not have access to the site where the sound is produced. The methodology presented here utilizes a binaural microphone where participants in the field, listen to a physical sound, and calibrate the binaural microphone by adjusting recording levels until the sound heard in their headphones matches the sound being produced live in the field. A second group of participants are presented with the same live physical sound but in an interior environment. Using the same setup, these participants compare the live sound with the recorded and reproduced sound calibrated by the in-field participants. Participants in this second group empirically evaluate the similarity of the reproduced sound to the live physical sound in the interior environment.

Session 4pBAa

Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Inverse Problems in Biomedical Ultrasound II

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Kang Kim, Cochair
Medicine, University of Pittsburgh, 950 Scaife Hall, 3550 Terrace Street, Pittsburgh, PA 15261

Contributed Papers

1:00

4pBAa1. Photoacoustic tomography in a clinical linear accelerator for quantitative radiation dosimetry. David A. Johnstone (Radiation Oncology, Univ. of Cincinnati, 3960 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, johnstdt@mail.uc.edu), Michael T. Cox (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Dan Ionascu, Michael A. Lamba (Radiation Oncology, Univ. of Cincinnati, Cincinnati, OH), Charles L. Dumoulin (Imaging Res. Ctr., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Cancer is the second leading cause of death in the United States. Approximately half of all cancer patients receive radiation therapy, in which linear accelerators are used to deliver high doses of x-ray radiation to tumors, inducing cell death. X-ray energy deposition causes pressure changes that produce acoustic signals due to the photoacoustic effect. Here, clinical x-ray beams were directed at test objects made of antimonial lead and other metallic materials within a water tank. Photoacoustic signals were measured using a calibrated broadband hydrophone and validated using simulations in k-Wave. Linear and two-dimensional synthetic apertures were formed by mechanically scanning the x-ray source and test object within a single plane. Tomographic images of test objects, reconstructed from measured photoacoustic signals, show good agreement with object geometry. X-ray doses incurred by the test objects are mapped based on the reconstructed acoustic pressure sources and Grüneisen parameter of the material employed. Potential applications to *in vivo* dosimetry for x-ray and proton therapy, potentially enabling safer and more effective treatments, are discussed.

1:15

4pBAa2. Comparisons of inverse and forward problem approaches to elastography. Siavash Ghavami, Saba Adabi (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., 200 First St. SW, Rochester, MN 55902, roudsari.seyed@mayo.edu), Olalekan Babaniyi (Civil and Environ. Eng., Duke Univ., Durham, NC), Azra Alizad (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN), Wilkins Aquino (Civil and Environ. Eng., Duke Univ., Durham, NC), and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN)

We present a full-wave inversion approach with total variation regularization for elastography. The proposed method is based on the minimization of an error in constitutive equations functional augmented with a least squares data misfit term referred to as MECE for "modified error in constitutive equations." The main theme of this paper is to demonstrate several key strengths of the proposed method on experimental data. In addition, some illustrative examples are provided where the proposed method is compared with a common shear wave elastography (SWE) approach. To this end, ultrasonically tracked displacement data from an acoustic radiation force (ARF) pulse are used in different phantoms including phantom with layered inclusion and triangle inclusion. The results indicate that the MECE approach can produce accurate shear modulus reconstructions in comparison with SWE, especially around the sharp edges in the layered and triangle inclusions. We compare shear modulus reconstruction using MECE and SWE with original inclusion shapes using two-dimensional normalized zero mean cross correlation, edge preservation index and dice coefficient similarity index. [Work supported by NIH Grant R01 CA174723.]

1:30

4pBAa3. Repeatability of linear and nonlinear quantitative compression elastography in the breast. Paul E. Barbone, Daniel Gendin (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu), Yuqi Wang (Univ. of Wisconsin, Madison, WI), Rohit Nayak (Mayo Clinic, Rochester, New York), Assad Oberai (Univ. of Southern California, Los Angeles, CA), Timothy J. Hall (Univ. of Wisconsin, Madison, WI), Azra Alizad, and Mostafa Fatemi (Mayo Clinic, Rochester, MN)

Compression elastography allows the precise measurement of large deformations of soft tissue *in vivo*. From a measured large deformation, an inverse problem for both the linear and nonlinear elastic moduli distributions can be solved. As part of a larger clinical study to evaluate NEMs in breast cancer, we evaluate the repeatability of linear and nonlinear modulus maps from repeat measurements. Within the cohort of 31 subjects scanned to date, several had repeated scans. These repeated scans were processed to evaluate NEM repeatability. *In vivo* data were acquired by a custom, digitally controlled, uniaxial compression device with force feedback. RF-data were acquired using plane wave imaging, at a frame-rate of 200 Hz, with a ramp-and-hold compressive force of 8N, applied at 8 N/s. A two-dimensional (2D) block-matching algorithm was used to obtain sample-level displacement fields which were then tracked at subsample resolution using 2D cross correlation. Linear and nonlinear elasticity parameters in the Blatz model of tissue elasticity are estimated using iterative optimization. Repeatability between both modes and elastic modulus maps is measured and compared. Preliminary results indicate that when images are acquired in the same region of tissue, the modulus maps are consistent. [Work supported by NIH R01CA195527.]

Session 4pBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics II

Jonathan Mamou, Cochair
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Contributed Papers

1:55

4pBAb1. Gas stabilizing titanium dioxide nanocones against desmoplastic cancer by ultrasound cavitation induced tumor penetration and sonodynamic therapy. Reju G. Thomas and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technol Univ., 62 Nanyang Dr., Block N1.2, 01-06, Singapore 637459, Singapore, jameskwan@ntu.edu.sg)

Sonodynamic therapy is an emerging technique for treating tumors by utilising ultrasound mediated reactive oxygen species (ROS) production from sonodynamic agents. Here, we have manufactured titanium dioxide nanocones (TDN) for local on-demand ROS generation. These nanocones nucleate inertial cavitation during exposure to therapeutic ultrasound. Furthermore, inertial cavitation enhances the penetration of the TDN into tissue. The particles were synthesized by a hydrothermal method in the presence of 1,6-hexanediamine as stabiliser. Electron microscopy images confirm the formation of nanocone structures with a size of 300 nm (and confirmed with dynamic light scattering). The TDN displayed an inertial cavitation threshold of 1.9 MPa for a 0.5 MHz ultrasound transducer at 5 mg/ml concentration. We also show that FITC conjugated TDN penetrated 2% agarose mold upto 1.2 cm distance after exposure to ultrasound for 10 min. Finally, a ROS release profile of TDN under ultrasound exposure was established using ROS sensor 1,3-diphenylisobenzofuran (DPBF). After 15 min of exposure to high intensity focused ultrasound (0.5 MHz center frequency) time, TDN in the presence of DPBF showed a significant decrease in UV absorbance compared to control, verifying that ROS were generated under ultrasound exposure. TDN opens up the potential for targeted sonodynamic cancer therapy.

2:10

4pBAb2. Acoustic microstreaming due to an oscillating contrast microbubbles near a substrate: Velocity, vorticity and closed streamlines. Nima Mobadersany and Kausik Sarkar (George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu)

Intravenously injected microbubbles used as ultrasound contrast enhancing as well as drug delivery agents are encapsulated by a nanometer-thick layer of lipids, proteins, or polymers to stabilize them against premature dissolution. Here, acoustic microstreaming due to an oscillating microbubble, either coated or free, responsible for sonoporation and other bioeffects is analytically investigated. The detailed flow field is obtained, and the closed streamlines due to the ring vortex are plotted in both Eulerian and Lagrangian descriptions. Analytical expressions are found for the ring vortex showing that its length depends only on the separation of the microbubble from the wall and the dependence is linear. The circulation as a scalar measure of the vortex is computed quantitatively identifying its spatial location. The functional dependence of circulation on bubble separation and coating parameters was shown to be similar to that of the shear stress. [Work supported partially by NSF CBET 1602884 and GWU.]

2:25

4pBAb3. Time-dependent nanobubble stability: Correlating bubble size and concentration with ultrasound performance. Eric C. Abenojar, Christopher Hernandez (Dept. of Radiology, Case Western Reserve Univ., 2185 S Overlook Rd., Cleveland Heights, OH 44106, eca20@case.edu), Judith Hadley (Malvern Panalytical, Westborough, MA), Al C. De Leon (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), Robert Coyne (Malvern Panalytical, Westborough, MA), Michael C. Kolios (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada), and Agata Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH)

Lipid shell-stabilized nanobubbles (NB, <500 nm) are widely explored as next-generation contrast agents for diagnostic ultrasound (US) imaging and drug delivery. For a successful clinical translation, it is important to understand the factors which contribute to the stability and rate of signal decay from the NB over time. The small size and fragile nature of NB have limited the characterization of their stability to correlations with their loss of signal over time under US. Bubble oscillations in the acoustic field, however, can accelerate their dissolution process. In this study, the passive, non-acoustically driven dissolution of lipid-shelled, C_3F_8 NB, and the relationship between bubble size/concentration and US signal intensity were assessed. The change in the acoustic activity of NB over time was correlated with the changes in size and concentration of the buoyant (bubbles) and non-buoyant particle population, measured using a novel resonant mass measurement technique. Clinical US was used to measure signal enhancement at different time points in a tissue phantom (f = 12.0 MHz, MI: 0.29, 1 fp/s). Results demonstrate a clear nonlinear relationship between the rate of ultrasound signal decay and concentration. While US signal decayed significantly over time (from 0 to 5 h), bubble concentration did not change significantly. A statistically significant decrease in the NB diameter was observed 1 h after the NBs were prepared and isolated while no change in the size was observed between 1 and 5 h.

2:40

4pBAb4. Microbubble translation in plane-wave ultrasound transmission. Outi Supponen (Mech. Eng., Univ. of Colorado at Boulder, 1111 Eng. Dr., Boulder, CO 80309, outi.supponen@colorado.edu), Francesco Guidi (Information Eng., Univ. of Florence, Florence, Italy), Awaneesh Upadhyay (Mech. Eng., Univ. of Colorado, Boulder, CO), Hendrik J. Vos (Biomedical Eng., Erasmus MC, Rotterdam, The Netherlands), Mark Borden (Mech. Eng., Univ. of Colorado, Boulder, CO), and Piero Tortoli (Information Eng., Univ. of Florence, Firenze, Italy)

Primary radiation force is capable of translating microbubbles in the focal region of single-element and array ultrasound probes. This effect can be harnessed to enhance the contact between ligand-bearing microbubbles and targeted endothelium for applications in targeted drug delivery and ultrasound molecular imaging. In this study, displacements of lipid-coated

microbubbles associated with plane-wave transmission are investigated using the multi-gate Doppler approach, and compared with focused-wave transmission at equivalent peak negative pressures. In plane wave transmission, the radiation force is nearly uniform over the field of view and therefore allows for a more uniform translation of microbubbles compared to focused wave. Statistically determined median displacements are in good agreement with the axial and lateral ultrasound beamplots both in plane-wave and focused-wave transmissions, while peak microbubble displacements reveal a number of discrepancies. Distinct size-isolated microbubble populations (diameters 1–2 μm, 3–4 μm, 4–5 μm, 5–8 μm, and polydisperse) were tested, showing important differences in their displacements and a strong driving frequency dependence thereof. These findings help tune the ultrasound transmission parameters for uniform and size-selective microbubble translations.

2:55

4pBAb5. Microstructural anisotropy evaluation in trabecular bone structure using the mode-converted (longitudinal to transverse, L-T) ultrasonic scattering. Omid Yousefian (North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27606, oyousef@ncsu.edu), Hualong Du (North Carolina State Univ., Lincoln, NE), Timothy Horn, and Marie M. Muller (North Carolina State Univ., Raleigh, NC)

The mode-converted ultrasonic scattering method is utilized to characterize the structural anisotropy of a phantom mimicking trabecular bone, fabricated using metal additive manufacturing from a high resolution CT image of trabecular horse bone. A normal incidence transducer transmits longitudinal waves into the sample, while the scattered transverse signals are received by an oblique incidence transducer. Four L-T measurements are performed by collecting scattering from four directions. The results show that the L-T converted scattering amplitude is highly dependent on the microstructural anisotropy direction. The ratios of L-T converted amplitudes for measurements in different directions is calculated to characterize the anisotropy of sample. The results show that the anisotropy is changing along the sample, which coincides with simulation results previously obtained on the same structures, as well as with the anisotropy estimated using image processing of the CT scans. The anisotropy was shown to increase monotonously along the sample from 0.48 to 0.7 depending on the location. At the same time, the ratio of LT scattering amplitude measured in two perpendicular directions was shown to increase monotonously from 0.6 to 0.67. These results suggest the potential of mode-converted methods to assess the anisotropy of structures including trabecular bone.

3:10

4pBAb6. Evaluation of bone fracture healing in children using acoustic radiation force: Initial *in vivo* results. Siavash Ghavami, Adriana Gregory, Jeremy Webb (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., 200 First St. SW, Rochester, MN 55902, roudsari.seyed@mayo.edu), Max Denis (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Lowell, MA), Viksit Kumar (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), Todd A. Milbrandt, A. Noelle Larson (Orthopedic Surgery, Mayo Clinic College of Medicine & Sci., Rochester, MN), Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), and Azra Alizad (Dept. of Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN)

Vibrational characteristics of the bone are directly dependent on bone's physical properties. In this paper, a vibrational method for bone evaluation is introduced. We propose a new type of quantitative vibro-acoustic method based on acoustic radiation force of ultrasound for bone characterization in patients with fracture. In this method, we excite the clavicle and ulna by an ultrasound radiation force (URF) pulse. The URF pulse induces vibrations in the bone, resulting in an acoustic wave that is measured by a hydrophone placed on the skin. The resulting acoustic signals were used for wave velocity estimation based on cross-correlation technique. To further separate different vibration characteristics, we adopt a variational mode decomposition (VMD) technique to decompose the received signal into an ensemble of band-limited intrinsic mode functions, which allows analyzing the acoustic signals in terms of their constitutive components. We conducted a prospective study that included a total of 15 patients, 12 with clavicle fractures and 3 with ulna fractures. The contralateral intact bones were used as control. Statistical analysis demonstrated that fracture bones can be differentiated from intact bone with a detection probability of 80%. Also, we introduce a "healing factor" that quantifies the progress of healing in clavicle bones. Statistical analysis showed that healing factor can track the progress of healing in clavicle in 80% of fractures.

3:25

4pBAb7. Ultrasonic bone assessment using backscatter difference measurements at 1 MHz. Brent K. Hoffmeister, Evan N. Main, and Phoebe C. Sharp (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, hoffmeister@rhodes.edu)

There is interest in developing ultrasonic techniques that can be used to detect changes in bone caused by osteoporosis. One approach, called the backscatter difference technique, measures the power difference between two portions of a backscatter signal from cancellous bone. Previous laboratory studies have tested the technique using transducers with center frequencies > 2 MHz. The present study uses a 1 MHz transducer which may improve performance at central skeletal sites such as the hip and spine. Measurements were performed *in vitro* on 54 cube shaped specimens of cancellous bone from 14 human femurs using a broadband, single element 1 MHz transducer. Received backscatter signals were analyzed to determine the normalized mean of the backscatter difference (nMBD) which was computed by measuring the power difference between two gated portions of the backscatter signal in decibels and dividing by the gate separation in microseconds. Linear regression analysis found weak to moderate correlations ($0.13 \leqq R \leqq 0.66$) between nMBD and bone density, depending on which portions of the signals were analyzed. These results suggest that backscatter difference measurements using a 1 MHz transducer may be able to detect changes in bone caused by osteoporosis.

3:40–3:55 Break

3:55

4pBAb8. Development of high frequency ultrasound-based technique to increase cell permeability. Sangpil Yoon (Aerosp. and Mech. Eng., Univ. of Notre Dame, 151 Multidisciplinary Res. Bldg., Notre Dame, IN 46556, syoon4@nd.edu), Yingxiao Wang (BIoEng., Univ. of California, San Diego, La Jolla, CA), and K. K. Shung (Biomedical Eng., Univ. of Southern California, Los Angeles, CA)

Ultra-high frequency ultrasonic transducers have been developed by limited groups for cellular applications and high resolution imaging purposes. We have developed 150 MHz ultrasonic transducers with a focal size of smaller than 10 μm to increase permeability of cells to introduce macromolecules into cell cytoplasm. Cell-based therapy has enormous potential to treat neurodegenerative disease and cancer by engineering cells. One of the main challenges in cell-based therapy has been the safe intracellular delivery of macromolecules such as proteins and nucleic acids. We have developed a high frequency ultrasound-based technique for simultaneous and targeted single cell intracellular delivery of diverse types of macromolecules by increasing permeability of cell. High frequency ultrasound has a focus with area smaller than a single cell and enough focusing gain to directly disrupt cell lipid bilayer without microbubbles. Extremely thin layer of lithium niobite single crystal was used to generate 150 MHz sound waves. The transducer was integrated with microscope to apply acoustic pulses to increase permeability. CRISPR-Cas9, programmable gene editing tools, were delivered into single cells after the permeability was increased by high frequency ultrasound beam transmitted from the developed high frequency ultrasonic transducers. This study showed that the direct disturbance of cell membrane without microbubbles can be achieved by high frequency ultrasound for the safe delivery of macromolecules by increasing cell permeability.

4:10

4pBAb9. Standing acoustic waves in microfluidic channels for enhanced intracellular delivery of molecular compounds. Connor S. Centner (BioEng., Univ. of Louisville, 580 S Preston, Louisville, KY 40202, connor.centner@louisville.edu), Mariah C. Priddy (BioEng., Univ. of Louisville, Louisville, KY), and Jonathan A. Kopechek (BioEng., Univ. of Louisville, Louisville, KY)

Intracellular delivery of molecular compounds is required for many *in vitro* research applications. Ultrasound-induced cavitation has been shown to enhance intracellular delivery of molecular compounds via mechanisms that may include sonoporation or endocytosis. Recently, acoustofluidic approaches have been developed to utilize standing acoustic waves (SAW) for cell manipulation in microfluidic channels. In this study, the effect of SAW on fluorescein release from perfluorocarbon double emulsion droplets in microfluidic channels was explored. Vaporization of perfluorocarbon double emulsion droplets induced by SAW may potentially enhance intracellular delivery of molecular compounds. In this study, fluorescein-loaded double emulsions were passed through a microfluidic device and exposed to 8-MHz SAW. Fluorescein release was quantified by measuring the change in fluorescence of the supernatant before and after treatment. Treatment with SAW in microfluidic channels increased the fluorescence by 8.8-fold compared to the baseline level. Fluorescein release was also higher after treatment with SAW compared to samples that passed through the microfluidic device without exposure to SAW ($p = 0.03$). These results suggest that SAW and perfluorocarbon double emulsions in microfluidic channels could potentially enhance the efficiency and consistency of intracellular molecular delivery *in vitro*.

4:25

4pBAb10. Sounding out bacteria: Microstructural effects of therapeutic ultrasound on bacterial biofilms. Lakshmi Deepika Bharatula (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., Singapore, Singapore), Scott Rice, Enrico Marsili (Singapore Ctr. for Environ. Life Sci. Eng., Nanyang Technolog. Univ., Singapore, Singapore), and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., 62 Nanyang Dr., Block N1.2, 01-06, Singapore 637459, Singapore, jameskwan@ntu.edu.sg)

Treatment of chronic infections due to formation of bacterial biofilms are a huge risk due to the growing concerns with antimicrobial resistance. Biofilms grow in a complex and dynamic environment that weaken the effect of antimicrobials. Yet, the current strategy to tackle the problem is the development of novel drugs. However, the increasing prevalence of antimicrobial resistance suggests that an alternative treatment strategy without, or in synergy with, antibiotics is necessary to combat the biofilm infections. We and others have proposed high intensity focused ultrasound (HIFU) as a means to disrupt the biofilm matrix and improve therapy. Yet to date, there is limited knowledge on the cellular activity triggered by the biofilm-acoustic interactions. Here, we report the effect of HIFU at 500 kHz center frequency in absence of antibiotics or microbubbles on the microstructure of biofilms formed by *Pseudomonas aeruginosa*. Changes to the biofilm after acoustic exposure were characterized by confocal microscopy and electrochemical impedance spectroscopy. We observed a drop in the biomass at pressures where non-linear acoustics were dominant, and an increase in cellular activity. Our results suggest that there are acoustic bio-effects present in these bacteria that have not yet been reported.

4:40

4pBAb11. Therapeutic ultrasound-induced insulin release *in vivo*. Tania Singh (Biomedical Eng., The George Washington Univ., 800 22nd St. NW, Ste. 5290, Washington, DC 20052, taniasingh@gwu.edu), Ivan Suarez Castellanos (INSERM, Washington, District of Columbia), Diti Chatterjee Bhowmick (Biological Sci., The George Washington Univ., Washington, DC), Joshua Cohen (GW Medical Faculty Assoc., Washington, DC), Aleksandar Jeremic (Biological Sci., The George Washington Univ., Washington, DC), and Vesna Zderic (Biomedical Eng., The George Washington Univ., Washington, DC)

We have previously shown that therapeutic ultrasound is capable of stimulating insulin release from pancreatic beta cells, non-invasively, safely and effectively. The aim of this work is to conduct preliminary animal studies to evaluate the feasibility of controlled insulin release *in vivo* using therapeutic ultrasound. Wild type hIAPP+/+ white FVB mice were randomly assigned to either the ultrasound treatment group or the sham group. Mice in the ultrasound treatment group received one five-minute treatment of continuous 1 MHz ultrasound at 1 W/cm^2 . Blood samples were collected via tail nick immediately prior to ultrasound application and immediately after ultrasound application. The pancreas was excised for histological analysis using H&E staining. No gross damage—including any burns on the skin—in the treatment area were observed and there was no evidence of skin burning or internal damage of the abdominal organs, especially the pancreas, found during necropsy. As measured by ELISA, the experimental group treated with ultrasound exhibited an increase of 0.43 ng/ml in blood insulin concentration compared to a 0.60 ng/ml decrease in the control group after 5 min ($p < 0.01$). Our preliminary results show promise in the translational potential of therapeutic ultrasound in the treatment of type 2 diabetes. We expect that our approach, with careful selection of ultrasound parameters, may provide a safe, controlled and targeted stimulation of insulin release from the pancreatic beta cells.

4:55

4pBAb12. Development and characterization of acoustically responsive exosomes for simultaneous imaging and drug delivery applications. Jenna Osborn (George Washington Univ., Ste. 3000, 800 22nd St. NW, Washington, DC 20052, jennakosborn@gwu.edu), Jessica Pullan, James Froberg, Yongki Choi, Sanku Mallik (North Dakota State Univ., Fargo, ND), and Kausik Sarkar (George Washington Univ., Washington, DC)

Exosomes are naturally secreted bilayer vesicles ranging in size from 40 to 200 nm that play a critical role in cell-to-cell communications and protein and RNA delivery. Researchers have explored exosomes as potential drug delivery vehicles due to their natural morphology and small size. Here, for the first time, bovine milk derived exosomes have been modified to be acoustically responsive as potential ultrasound contrast agents or a drug carrier. The echogenic exosomes were formed through a freeze-drying process in the presence of mannitol. The size and morphology of the particles were assessed with a qNano^{TM} and atomic force microscopy (AFM). The ultrasound response of these particles was characterized through linear and nonlinear scattering behaviors. The presence of the echogenic exosomes enhances the scattered signal by 11.4 ± 6.3 dB. The stability of these particles under constant ultrasound exposure were assessed to be similar to that of echogenic polymersomes. The variation of mannitol concentration was assessed. To assess the imaging improvement of ultrasound imaging, the exosomes were injected through a tail vein in mice. The modification of the echogenic exosomes shows to have great promise as potential ultrasound contrast agents or ultrasound responsive drug delivery system.

5:10

4pBAb13. Therapeutic ultrasound for improving the tropical corneal delivery of macromolecules. Hanaa H. Almogbil, Camille Daszynski (School of Eng. and Appl. Sci., The George Washington Univ., Sci. & Eng. Hall, 800 22nd St. NW, Washington, DC 20052, Hanaa@gwu.edu), Erik A. Rodriguez (Dept. of Chemistry, The George Washington Univ., Washington, DC), Tania Singh (School of Eng. and Appl. Sci., The George Washington Univ., Washington, DC), Mary Ann Stepp (School of Medicine, The George Washington Univ., Washington, DC), and Vesna Zderic (School of Eng. and Appl. Sci., The George Washington Univ., Washington, DC)

The goal of this project is to facilitate the delivery of topical drugs into the cornea and anterior segment of the eye using therapeutic ultrasound which could present a promising treatment for keratoconus and other corneal diseases. Each cornea is dissected and placed in a diffusion cell. smURFP-blue, a blue fluorescent chromophore, was used as the drug. The experimental groups of corneas were treated with 1 and 0.8 W/cm^2 continuous ultrasound for 5 min at frequencies of 400 kHz and 600 kHz, respectively, then left in the diffusion cell for another 55 min. Fluorescence images of the fixed corneas were obtained to determine the relative amount of smURFP-blue that remained in the tissue. Safety of ultrasound application was tested by comparing the damage in the corneal layers. Spectroscopy measurements indicated no statistical difference in the presence of the chromophore in the receiver compartment in ultrasound- and sham-

treatment groups. Preliminary results showed greater fluorescent intensity in the cornea when smURFP-blue is delivered with ultrasound compared to smURFP-blue added without ultrasound. The histology studies did not show any significant damage in ultrasound-treated corneas. This work may allow for the development of an inexpensive, clinically applicable, and minimally invasive ultrasound method for corneal drug delivery.

5:25

4pBAb14. Optimization of molecular delivery to red blood cells using an ultrasound-integrated microfluidic system. Emily M. Murphy, Mariah C. Priddy (BioEng., Univ. of Louisville, 200 E. Shipp Ave., Louisville, KY 40208, emmurp09@louisville.edu), Brett R. Janis, Michael A. Menze (Biology, Univ. of Louisville, Louisville, KY), and Jonathan A. Kopechek (BioEng., Univ. of Louisville, Louisville, KY)

The shelf-life of donated red blood cells (RBCs) for transfusions is currently limited to six weeks when stored under refrigeration. This causes supply shortages worldwide and prevents transfusions in locations that lack access to cold-chain storage. Recently, a new approach to store RBCs as a dried powder at ambient temperature was developed. This method utilizes an ultrasound-integrated microfluidic platform to induce intracellular delivery of compounds that protect cells during desiccation and rehydration. The objective of this study was to detect cavitation emissions in order to optimize parameters for molecular delivery to RBCs in this system. Ultrasound was continuously generated in the microfluidic channels using an 8-MHz PZT plate and acoustic emissions were passively detected with an identical PZT plate aligned coaxially. Fluorescein and lipid-coated microbubbles were added to RBC solutions in order to nucleate cavitation and enhance intracellular molecular uptake as measured by flow cytometry. Increased levels of broadband emissions were detected at microfluidic flow rates associated with higher fluorescein delivery to RBCs. These results suggest that inertial cavitation plays an important role in enhancing molecular delivery to RBCs in the microfluidic channels. Optimization of this system may enhance delivery of protective compounds for long-term preservation of blood.

THURSDAY AFTERNOON, 16 MAY 2019 WILLIS, 1:30 P.M. TO 3:30 P.M.

Session 4pEA

Engineering Acoustics: General Topics in Engineering Acoustics: Characterization and Measurement

Matthew D. Guild, Cochair
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Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758

Contributed Papers

1:30

4pEA1. Modeling the vibration of a thin bar using SimScape. Carter J. Childs and Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ.University Park, PA 16802, cjc357@psu.edu)

The standard treatment of longitudinal and flexural vibrations of thin bars follow the methods described by Raleigh. The solution for longitudinal vibrations is completely analogous to electrical signals in electrical transmission lines. Thus, longitudinal mechanical vibrations can be modeled using lumped or distributed parameter analog circuits. The same is true of transverse vibrations of perfectly flexible strings. However, when bending stiffness is included for transverse vibrations, the simplicity of the transmission line analogies is not present. The authors are not aware of a lumped parameter model of the flexural vibration of a thin bar that includes the effects of bending stiffness. This paper presents a lumped parameter model for a thin bar that provides accuracy similar to that of a lumped parameter model of a flexible string. The differential length element of the bar is modeled in the same way as the differential element in the standard treatment of bra vibrations. The model will be demonstrated in the SimScape modeling software, though it can also be implemented in Modelica and possibly in some versions of SPICE.

1:45

4pEA2. Engine characterization using experimental method and prediction of insertion loss of the exhaust system. Manish Chhabra (Mech., Univ. of Cincinnati, 275N Marr Rd., Apartment 101, Columbus, IN 47201, chhabra.manish90@gmail.com)

The evaluation of the acoustic performance of an exhaust system at the design stage requires a correlated engine model and reasonably approximate input boundary conditions to simulate the end results, both of which are not easily available. It is known that the input boundary conditions for insertion loss analysis require two engine parameters, namely, source impedance and source strength spectra. This study describes experimental measurements for these parameters using in-duct measurement via the multi-load method for a six-cylinder diesel engine and calculation for insertion loss using GT-Power. The research discusses the approach taken to select the acoustic load cases considered for the multi-load method and then the execution of the test plan for different engine operating conditions The time domain data sets were processed to obtain the frequency spectra and was used to get the impedance of the acoustic load cases and finally the source impedance and source strength spectra for different engine operating conditions. The results obtained using all the acoustic load cases were optimized by filtering out unacceptable load cases and then re-evaluating the source characteristics to use them as input boundary conditions for insertion loss analysis. The analysis results were then compared to the experimental insertion loss.

2:00

4pEA3. Impedance measurements of acoustically nonlinear engine components. Eric Van horn, Luke Humphreys, and David E. Scarborough (Aerosp. Eng., Auburn Univ., 211 Davis Hall, AL 36832, hnh0004@auburn.edu)

Currently, the acoustic analysis of combustion systems generally consists of simplified impedance networks. For effective simulation, the impedance of these components must be known, for which there is a large gap in the current knowledge. This study examines the acoustic impedance for various different types of gas-turbine engine components, including acoustically nonlinear components such as swirlers and orifice plates. Experimental results are obtained using a specially designed impedance tube. Acoustic pressure and velocity measurements are made using a multiple microphone method. The impedance is then calculated using the reflection coefficient, and impedance subtraction used to isolate the test article. This method obtains accurate results within a relatively wide range of frequency values, 100–1500 Hz for this study. Amplitude sweeps conducted for each test article demonstrate the nonlinear aspects, if any, for each test article. These results are then compared to acoustic simulations conducted in COMSOL to assess the capabilities and shortcoming of COMSOL's linear acoustic package to provide predictions of the acoustic impedance of these components. The results for this are intended to provide valuable data for modeling the acoustics of combustion systems, as well as demonstrate an effective method for obtaining impedance data for various acoustic components.

2:15

4pEA4. Initial results in designing an acoustic sound simulator for heavy equipment. Nathaniel Wells, Scott D. Sommerfeldt, and Jonathan Blotter (Brigham Young Univ., N308 ESC, Provo, UT 84602, nateswells@gmail.com)

This paper focuses on the initial development of a vehicle cab sound simulator. This sound simulator has two objectives. First, it can be coupled with a visual simulator and used for operator training. Second, it will be used by designers such that when structural modifications are made to the vehicle, the acoustic response and sound quality in the cab can be predicted. To begin to understand and implement this simulator, transfer functions were measured for several structures progressing from simple to complex and used to generate a simulated signal. Post-processing techniques were used to improve the overall quality of the simulated responses. Similarly, the structures were recreated in a numerical software package, where the transfer functions were calculated numerically and used to generate simulated responses. These signals were compared to the measured response of the system and auralized to determine the effectiveness of the simulation.

2:30

4pEA5. On a cooling speed comparison a sound fire extinguisher with the blade. Bong Young Kim and Myungjin Bae (Commun. Eng., Soongsil Univ., 21-1, Garak-ro 23-gil, Songpa-gu, #203, Seoul 05669, South Korea, bykim8@ssu.ac.kr)

Sound Fire Extinguisher, which is actively studied at Sori Sound Engineering Research Institute (SSERI), is a new type of extinguishment facility that can be used for suppression and prevention of conflagration in various environments. Sound Fire Extinguisher uses acoustic lens to minimize the attenuation of sound energy and transfer energy to the target point. It can prevent conflagration by lowering ambient temperature even before conflagration. In this study, we experimented to see if the Sound Fire Extinguisher could prevent conflagration by lowering the ambient temperature. Experimental results show that when the Sound Fire Extinguisher sound component of the same wind speed is supplied, the heated tableware is cooled by 10%–20% faster than the wind speed of 2 m/s. These results show that the Sound Fire Extinguisher can be used to prevent conflagration, since the sound component of the Sound Fire Extinguisher itself promotes the surrounding thermal dissipation to cool quickly.

2:45

4pEA6. Design and characterization of tabletop anechoic and sonication chambers. Olivia Ott, Shane Lani, and Jim Sari (Johns Hopkins Univ. Appl. Phys. Lab. 11100 Johns Hopkins Rd., Laurel, MD 20723, olivia.ott@jhuapl.edu)

Access to anechoic chambers for acoustic testing can be an expensive and limited resource for acousticians. The purpose of this work was to develop small scale, portable anechoic and sonication chambers that could be used for rapid and simultaneous testing. Using consumer off the shelf (COTS) products, low cost anechoic and sonication chambers were designed and built. Upon completion the prototypes were characterized for their acoustic performance. Both the anechoic and sonication chambers were characterized by their ability reduce ambient noise. The characterization was completed by measuring transmission loss (TL) through the chamber walls. Second, the sonication chamber was tested for spatial sound pressure level (SPL) variation. The procedure for spatial testing inside the chamber included measuring SPL along a horizontal stage using mini microphones. The performance and characterization of these chambers will be presented and discussed.

3:00

4pEA7. Absorption measurement of acoustic materials using a spherical microphone array. Samuel Dupont (LAUM / GAUS, Universite du Mans, Ave. Olivier Messiaen, Le Mans 72085, France, samuel.dupont@univ-lemans.fr), Manuel Melon (LAUM, Le Mans, France), and Alain Berry (GAUS, Sherbrooke, QC, Canada)

Standardized methods for the sound absorption measurement are restricted to normal or diffuse incidence. The capture of the sound field over a plane material sample using a spherical microphone array has already been proposed as a solution to measure surface impedance or sound absorption. However, the proposed solutions use either modal beamforming or a direct computation of the pressure on the sphere. In this presentation, a method which uses the spherical harmonics expansion to calculate the absorption using the spherical holography principle is proposed. It allows to measure the characteristics of the material without relying on an accurate definition of the sound scene such as the exact position of the source but only on the distance of the spherical microphone to the material. Simulation and experimental result of surface impedance and sound absorption will be presented as well as comparisons with another approach from the literature based on sparse array processing of spherical antenna.

3:15

4pEA8. Ultimate wave decomposition method for characterization of symmetric and asymmetric materials. Reza Ghaffarivardavagh, Rui Wu (Mech. Eng., Boston Univ., 8 saint Mary St., Photonic Bldg., Rm. 832, Boston, MA 02215, ghaffari@bu.edu), Stephan Anderson (Boston Univ. Medical Ctr., Boston, MA), and Xin Zhang (Mech. Eng., Boston Univ., Boston, MA)

To date, several approaches leveraging impedance tubes have been proposed to retrieve the complex acoustic reflection and transmission coefficient through the materials. In these approaches, commonly, accurate derivation of the reflection and transmission relies on the several assumptions such as material symmetry or the type of the terminations (load) used in the impedance tube. Herein, we have proposed an Ultimate wave decomposition approach (UWD) that may be employed for the precise derivation of the reflection and transmission coefficients, regardless of the termination type. Moreover, it is demonstrated that by employing the UWD method, utilizing single arbitrary termination, symmetric material, and two arbitrary terminations, asymmetric material can be fully characterized. The performance of the presented methodology has been compared experimentally with four different methods with respect to both symmetric and asymmetric cases, from which the superior performance of the UWD method is demonstrated. UWD represents a highly practical approach in which reflection and transmission and consequently acoustical properties of the material, can be precisely measured and calculated using a very simple procedure.

Session 4pNS

Noise, Signal Processing in Acoustics, and Psychological and Physiological Acoustics: Advances and Applications in Sound Quality Metrics

S. Hales Swift, Cochair
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Patricia Davies, Cochair
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Chair's Introduction—1:05

Invited Papers

1:10

4pNS1. The loudness model used in ISO532-3: Development, evaluation and prospects. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

ISO 532-2 is based on the loudness model for stationary sounds described by Moore and Glasberg [*JASA* (2007)]. This model is similar to that in ANSI S3.4-2007, except that the model in ISO 532-2 incorporates binaural inhibition: a strong input to one ear in a given frequency region reduces the effective level of a weaker input to the other ear in nearby frequency regions. ISO 532-3 has been proposed as an extension to ISO 532-2 to deal with time-varying sounds. It generates predictions of short-term loudness, the loudness of a short-segment of sound such as a word in a sentence or a single note in a piece of music, and of long-term loudness, the loudness of a longer segment of sound, lasting 1–5 s, such as a whole sentence or a musical phrase. The model gives reasonably accurate predictions of the overall loudness of technical sounds (e.g., factory noises), of speech whose dynamic range has been compressed or expanded, and of sounds whose time pattern and spectra differ at the two ears. However, the model needs to be extended to generate predictions of the overall loudness impression of a sound environment over a period of several minutes to an hour.

1:30

4pNS2. Rapid calculating of loudness according to ANSI S3.4-2007 with the Glasberg and Moore 2002 extension to time-varying signals in MATLAB. S. Hales Swift (Energy Systems Div., Argonne National Lab., N221 ESC, Provo, Utah 84602, hales.swift@gmail.com), and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The ANSI S3.4-2007 standard gives a method for calculating the predicted loudness of stationary sounds for an average listener. Glasberg and Moore (2002) provide an extension of the method to time-varying sounds. The mathematical structure of the excitation in the loudness calculation is amenable to significant acceleration in MATLAB by expressing portions of the calculation, notably those representing the cochlear filtering process, in terms of matrices. Thus, procedures to achieve rapid processing of loudness are set forth. Possible extensions of this approach to other metrics within the same family are considered.

1:50

4pNS3. The description of fan noise by indexes based on the specific loudness. Stephan Töpken and Steven van de Par (Acoust., Univ. of Oldenburg, Carl-von-Ossietzky-Str. 9-11, Oldenburg 26129, Germany, stephan.toepken@uni-oldenburg.de)

In a previous study of the authors, a broad variety of fan sounds that were equalized in overall A-weighted level was rated in listening experiments with a semantic differential. The factor analyses of the ratings indicated six perceptual dimensions and five groups of sounds, which shows the rich variety of sound characteristics covered by the tested fan sounds. The results showed that the groups of pleasant and unpleasant sounds differed mainly with respect to the first three perceptual dimensions, "pleasant," "humming/bass," and "shrill." An analysis of the specific loudness according to the DIN 45631 standard revealed systematic differences in the specific loudness patterns for the different groups of fan sounds. It was possible to define two psychoacoustic indexes that correlate highly with the factor values of the three most important perceptual dimensions of fan noise. The most important index, N_{ratio}, relates the amount of loudness resulting from low mid-frequency content between 2 Bark and 5 Bark to the loudness from high frequency content above 10 Bark. The identified boundaries of the frequency ranges employed in the indexes are in good agreement with those found for air conditioning noise, air cleaners and in the context of sound masking in offices.

2:10

4pNS4. Psychoacoustic Roughness Standard. Roland Sottek and Julian Becker (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

Roughness of acoustic signals has been a focus of sound design for many years. A rough sound can contribute to a sporty character of an engine, but also to a strong annoyance. It is desired that a sound fits the product. This goal should be achievable as early as possible in the development process. For this purpose, perceptual evaluations in combination with model calculations and simulation tools must be used. Existing roughness calculation models work well for synthetic signals such as modulated tones and noise signals. However, the roughness prediction is much more challenging for technical sounds because of the more complex spectral and temporal patterns. Although the consideration of roughness is very common in practice, there is still no standardized roughness calculation method. This paper describes a method that is based on a model of human hearing according to Sottek. It has been optimized for non-linear processing and the weighting of the modulation spectra. Additionally, a proposal for calculating a binaural single value of roughness is given. This model allows to predict the perceived roughness very well. The standardization of this roughness calculation method is planned both as a German standard (DIN) and as part of ECMA-74.

2:30

4pNS5. Inconsistencies between the predicted qualities of enhanced speech signals from two objective metrics. Zhuohuang Zhang and Yi Shen (Speech and Hearing Sci., Indiana Univ. Bloomington, 200 S Jordan Ave., Bloomington, IN 47405, shen2@indiana.edu)

Objective speech-quality metrics have been used widely as a tool to evaluate the performance of speech enhancement algorithms. Two widely adopted metrics are Perceptual Evaluation of Speech Quality (PESQ) and Hearing-Aid Speech Quality Index (HASQI). While PESQ is based on a highly-simplified phenomenological model of auditory perception for normal-hearing listeners, HASQI contains processing steps that represent the physiology of the auditory periphery and is able to capture the perceived speech quality from hearing-impaired listeners. In the current study, the performance of deep-learning-based speech enhancement algorithms was evaluated using the two objective metrics. The algorithms were implemented so that the audio features were represented in either a linear frequency scale or a nonlinear frequency scale (i.e., Mel scale). Higher speech quality for linear- than Mel-scale processing was predicted by HASQI, but PESQ was less sensitive to the difference in frequency scale. To resolve the discrepancies between the two metrics, a behavioral experiment was conducted following the ITU recommended procedure for assessing speech quality (ITU-R BS.1534-1). Listeners strongly preferred the enhanced speech using linear- over Mel-scale processing, which is consistent with predictions from HASQI. The sources of the discrepancies between PESQ and HASQI were also explored via further acoustic analyses.

2:50

4pNS6. Reliability and validity of sound quality metrics versus objectivity. Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

Often it is mentioned to get an objective measurement of the subjective evaluation. The result is a sound quality metric. The terms subjective and objective are traditionally associated with aspects related to human perception and physical measures respectively. Objectivity describes the independency of test results from the respective researcher, reliability considers whether the same results would be achieved, if the research procedure would be repeated, validity means whether a method measures what it is intended to measure. Physical measurements could have a high reliability but low validity, whereas perceptual measurements often possess a relatively low reliability but relatively high validity. For a sound quality metric, the use of predictors supported by theoretical considerations and plausibility is very important. Furthermore, a robustness analysis is needed indicating that the metric is not very susceptible to choice of input data. Be aware, the use of statistics does not replace thinking; the use of predictors must be plausible or needs theoretical background. Examples of typical sound quality metrics will be given.

3:10–3:25 Break

3:25

4pNS7. Identification of perceived sound discomfort contributed from partially correlated vibration and noise sources in vehicles. Yu Huang and Weikang Jiang (Shanghai Jiao Tong Univ., Dongchuan Rd. 800 Jidong A819, Minhang, Shanghai 200240, China, yu_huang@sjtu.edu.cn)

The interior noise sources are often complex in vehicles, including not only structure-borne and air-borne noise sources but also vibration sources. These sources may be partially correlated and cannot be calculated using traditional methods, e.g., transfer path analysis and operational path analysis. On the other hand, it is necessary to study the sound quality of the vehicle interior noise to improve the comfort of drivers and passengers in vehicles. An operational partial singular value decomposing method together with sound quality analyses was employed in this study to determine the influence of various partially correlated sources with the perceived discomfort of the subject. The vibration and noise in vehicles were measured in a car when it was running on asphalt, concrete, gravel and bumpy roads. Thirty subjects used the absolute magnitude estimation method to rate the discomfort produced by noise stimuli. A discomfort model was proposed based on the relations between subjective magnitudes and the objective parameters of noise (i.e., the SPL, loudness, roughness, sharpness, and articulation index). The contributions of various vibration and noise sources to the vehicle discomfort were predicted well by the operational partial singular value decomposing method based on the discomfort model.

3:45

4pNS8. Sound quality metric indicators of rotorcraft noise annoyance using multilevel regression analysis. Matthew Boucher, Siddhartha Krishnamurthy, Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681-2199, matthew.a.boucher@nasa.gov), and Stephen A. Rizzi (AeroAcoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

It is hypothesized that sound quality metrics, particularly loudness, sharpness, tonality, impulsiveness, fluctuation strength, and roughness, could all be possible indicators of the reported annoyance to helicopter noise. To test this hypothesis, a psychoacoustic test was recently conducted in which subjects rated their annoyance levels to synthesized helicopter sounds [Krishnamurthy, InterNoise2018, Paper 1338]. After controlling for loudness, linear regression identified sharpness and tonality as important factors in predicting annoyance, followed by fluctuation strength. Current work focuses on multilevel regression techniques in which the regression slopes and intercepts are assumed to take on normal distributions across subjects. The importance of each metric is evaluated one-by-one, and the variation among subjects is evaluated using simple models. Then, more complete models are investigated, which include the combination of selected metrics and random effects. While the conclusions from linear regression analysis are affirmed by multilevel analysis, other important effects emerge. In particular, a random intercept is shown to be more important than a random slope. In this framework, the relative importance of sound quality metrics is re-examined, and the potential for the modeling of human annoyance to helicopter noise based on sound quality metrics is explored.

4:05

4pNS9. Identifying metrics to predict annoyance due to Mach-cutoff flight ground signatures. Nicholas D. Ortega (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, njo5068@psu.edu), Michelle Vigeant (Acoust., The Penn State Univ., State College, PA), and Victor Sparrow (Acoust., The Penn State Univ., University Park, PA)

Theoretically, Mach-cutoff flight under ideal atmospheric conditions could lead to boomless supersonic flight observed under the flight path on the ground. Such ideal atmospheric conditions refract the sonic boom waves upwards at the caustic line, so they do not reach the ground. This presentation describes the perception of the evanescent sound field below the flight path. The work investigates perceptual attributes and metrics related to these unique sounds. Annoyance and three other perceptual factors ("Thunderous," "Rumbly," and "Swooshing") were analyzed through subjective testing using pair-wise comparison. Stimuli used were from recordings made during NASA's "Farfield Investigation of No-boom Thresholds" (FaINT). Linear regression with principal component analysis indicated which perceptual factors contribute to annoyance, and stepwise regression identified candidate metrics for predicting annoyance. Traditional loudness metrics (i.e., weighted Sound Exposure Level) were analyzed alongside sonic-boom specific metrics (i.e., Perceived Loudness) and sound quality metrics (i.e., Sharpness). [Work supported by the U.S. Federal Aviation Administration Office of Environment and Energy through ASCENT, the FAA Center of Excellence for Alternative Jet Fuels and the Environment, Project 42 through FAA Award No. 13-C-AJFE-PSU under the supervision of Sandy Liu. Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

4:25

4pNS10. Comparison of sonic boom noise metrics from predictions and measurements under low atmospheric turbulence conditions. Alexandra Loubeau and William Doebler (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Six noise metrics have been identified as candidates for quantifying ground sonic boom levels from overflight of supersonic aircraft. Each of these metrics (PL, ASEL, BSEL, DSEL, ESEL, and ISBAP) has previously been investigated in meta-analyses using laboratory study data corresponding to perception of sonic booms in outdoor and indoor environments. These metrics are now computed and analyzed for a set of recorded outdoor sonic boom signatures under low atmospheric turbulence conditions. Predictions of the ground signatures are also computed, without inclusion of turbulence effects, and metrics are compared between measurements and predictions. Metrics least sensitive to atmospheric turbulence effects are identified as potentially more robust for quantifying the sonic boom level from a supersonic aircraft.

Contributed Papers

4:45

4pNS11. Stevens perceived levels of common impulsive noises, sonic booms, and sonic thumps. William Doebler and Jonathan Rathsam (NASA Langley Res. Ctr., M.S. 462, NASA Langley Res. Ctr., Hampton, VA 23681, william.j.doebler@nasa.gov)

NASA's X-59 Quiet Supersonic Technology low boom flight demonstrator aircraft is being designed to produce a shaped sonic thump of 75 dB Perceived Level (PL) at the ground. The PL metric was chosen because it correlates well with human perception of sonic booms both outdoors and indoors. Members of the public often ask how loud 75 dB PL is. To communicate this level in terms of more familiar sounds, a PL reference scale was developed. Common impulsive sounds were recorded, and their PLs were computed. Some of the various impulsive sounds include distant thunder, basketball bounces, and car door slams (79, 81, and 89 dB PL, respectively). Concorde's 105 dB PL traditional N-wave sonic boom is also included in the reference scale. Additionally, the impulsive sounds' energy spectral densities and sone spectra are compared to that of a simulated X-59 ground waveform.

5:00

4pNS12. Developing a tone standard for air-conditioning and refrigeration equipment. Derrick P. Knight (Ingersoll Rand, 3600 Pammel Creek Rd., La Crosse, WI 54601, derrick.knight@irco.com)

AHRI Technical Committee on Sound is continuing the redevelopment of standard AHRI 1140—Sound Quality Evaluation Procedures for Air-Conditioning and Refrigeration Equipment. We are currently evaluating the feasibility of harmonizing this standard with a current ASHRAE funded study whose goal is to determine the threshold of annoyance due to tones in HVAC equipment. However, from a manufacturer's perspective, it is very difficult to accommodate a metric measured in the listener's space. Additionally, sound power test methods for HVAC equipment allow testing in a reverberant field, which poses significant challenges to measuring tones. This presentation will provide an update in regards to the development for

standard AHRI 1140 and will seek feedback from consultants and academics.

5:15

4pNS13. Effect of trading-off office background and intermittent noise levels on performance, annoyance, distraction, and stress. Martin S. Lawless, Zane T. Rusk, Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, lawles@cooper.edu), and Andrew Dittberner (GN Hearing, Glenview, IL)

In open-plan offices, work performance is affected by the acoustic environment, which includes steady-state broadband noise and intermittent, occupancy-generated noise. High levels of broadband noise (e.g., HVAC noise) can mask intermittent sounds to reduce distraction, but risk causing fatigue and other noise-related symptoms that may be detrimental to performance. In this study, the impact of the acoustic environment on work performance was investigated by adjusting the relative levels of both broadband and intermittent noise. Participants were exposed to four different acoustic environments, either starting with high background noise and low intermittent levels or vice versa. While in each background condition, the subjects performed four cognitive tasks that evaluated memory, attention, reasoning, and planning skills, respectively. Heart rate variability and electrodermal activity (EDA) were measured to gauge arousal (stress levels) in each environment. After each exposure, participants were asked to rate annoyance, distraction, fatigue, and stress, among other subjective attributes. The EDA and ratings of distraction significantly increased as the intermittent noise levels increased, while noise annoyance ratings were consistent across each background condition. Additionally, performance on the cognitive tasks was impacted by the order in which the participants experienced the acoustic environments.

THURSDAY AFTERNOON, 16 MAY 2019 JONES, 1:00 P.M. TO 3:30 P.M.

Session 4pPAa

Physical Acoustics and Signal Processing in Acoustics: Infrasound II

Roger M. Waxler, Cochair
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Philip Blom, Cochair
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Contributed Papers

1:00

4pPAa1. Infrasound propagation in multiple-scale random media using generalized polynomial chaos. Alexandre GOUPY (CMLA, ENS Paris-Saclay, CMLA, ENS Paris-Saclay, Cachan, France, alexandre.goupy@gmail.com), Christophe MILLET (CEA, DAM, DIF, Arpajon, France), and Didier LUCOR (LIMSI CNRS, Orsay, France)

Infrasound propagation in realistic environments is highly dependent on the information to specify the waveguide parameters. For real-world applications, there is considerable uncertainty regarding this information, and it is more realistic to consider the wind and temperature profiles as random functions, with associated probability distribution functions reflecting phenomena that are filtered out in the available data. Even though the numerical methods currently-in-use allow accurate results for a given atmosphere, high dimensionality of the random functions severely limits the ability to compute the random process representing the acoustic field, and some form of sampling reduction is necessary. In this work, we use polynomial chaos (gPC)-based metamodels to represent the effect of large-scale atmospheric variability onto the acoustic normal modes. The impact of small-scale atmospheric structures is modelled using a perturbative approach of the coupling matrix. This multi-level approach allows to estimate the statistical influence of each mode as the frequency varies. An excellent agreement is obtained with the gPC-based propagation model, with a few realizations of the random process, when compared with the Monte Carlo approach, with its thousands of realizations. Furthermore, the gPC framework allows computing easily the Sobol indices without supplementary cost, which is essential for sensitivity studies.

1:15

4pPAa2. Similarities and differences in infrasound propagation effects between arctic and temperate environments. Michelle E. Swearingen (U.S. Army ERDC, Construction Eng. Res. Lab., P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Sarah McComas (U.S. Army ERDC, Vicksburg, MS), D. K. Wilson, and Vladimir Ostashev (U.S. Army ERDC, Hanover, New Hampshire)

Meteorological conditions in an arctic environment differ significantly from those in a temperate environment. Atmospheric phenomena particular to polar regions, including wind patterns such as the polar vortex and low-level jets above strongly stable layers, strong temperature and humidity gradients, and density currents, could have unique impacts on infrasound propagation that are not observed in temperate locations. In this study, parabolic-equation simulations of sound propagation are performed using measured meteorological conditions for summer and winter conditions in temperate and arctic locations. The similarities and differences in environmental conditions between these two locations and their relative impact on the predicted transmission loss are examined. For summer conditions, a comparison to measured data from explosive sources is performed for both temperate and arctic locations.

1:30

4pPAa3. Dependence of Rayleigh integral predictions on the ground motion model for acoustic signals produced by underground explosions. Philip Blom (Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, pblom@lanl.gov)

Seismoacoustic signatures are produced by above- and below-ground explosions and are often observed at local, regional, and global distances. In the case of an underground explosion, seismic waves propagate to the surface and produce acoustic signatures via pumping of the atmosphere by the ground motion that can be predicted using a Rayleigh integral analysis. Acoustic signature predictions will be discussed and compared with observations from the Source Physics Experiment (SPE) in two scenarios. First, a rigid piston model of the ground motion will be highlighted as a first order model. Second, a more realistic model treating the ground as an elastic medium with finite compressional wave speed will be developed and discussed to demonstrate how such a model changes predicted acoustic signals at local distances. Acoustic signals predicted using each methodology will be compared with observations from SPE to identify how characteristics of the acoustic signal can be leveraged to improve characterization of the underground explosive source.

1:45

4pPAa4. Validation of a generalized least squares beamformer for infrasonic data analysis. Fransiska K. Dannemann, Philip Blom, and Omar Marcillo (Earth and Environ. Sci., Los Alamos National Lab., P.O. Box 1663, MS D446, Los Alamos, NM 87545, fransiska@lanl.gov)

Current seismoacoustic signal detectors including the traditional F-detector, the Progressive Multi-Channel Correlation detector (PMCC), and the adaptive F-Detector (AFD) statistically separate signals of interest from noise based upon a user-defined threshold, however, in regions of high background noise or in the presence of multiple transient signals, a signal's SNR decreases and is often missed by the detector. The adaptive F-detector addresses this problem of coherent noise across array elements by re-mapping a noise threshold over a user-defined window. While application of the adaptive F-detector successfully reduces false detection rates attributed to coherent noise across array elements, the detector is applied post-processing following array analysis using a standard (Bartlett) beamforming approach. Processing of low SNR infrasonic signals can further be enhanced through the application of a generalized least squares (GLS) approach to beamforming which adaptively accounts for background noise characteristics. A characterization of background noise environments will be presented, along with the statistical significance of enhanced detection capabilities compared to traditional beamforming approaches.

2:00

4pPAa5. An investigation of the scattering of infrasonic acoustic waves by turbulent fluctuations generated by the breaking of gravity waves. Roberto Sabatini, Jonathan B. Snively, and Michael P. Hickey (Embry-Riddle Aeronautical Univ., 1400 Hancock Blvd., Daytona Beach, FL 32114, roberto87sabatini@gmail.com)

Infrasonic waves generated below and above the Earth's surface can travel up to ionospheric heights and also reach very large radial distances, spanning from hundreds to thousands of kilometers. As a result, the signals recorded at ground level far from the source location and at high altitudes are strongly influenced by the spatial and temporal variations of the temperature and winds. In most propagation models, acoustic waves are treated as perturbations of a stationary mean atmosphere, which varies only along the vertical coordinate. Hence, horizontal and temporal small-scale fluctuations of temperature and winds induced by gravity waves are inherently excluded by such methods. The objective of the present work is two-fold. First, a model based on the compressible unsteady Navier-Stokes equations, is applied to simultaneously investigate the propagation and breaking of gravity waves and the propagation of infrasonic waves (here emphasizing frequencies in the range [0.001,0.1] Hz) through their induced fluctuations. Second, simulations are performed to investigate the effects of small-scale turbulent inhomogeneities on infrasonic recordings at the ground and within the thermosphere-ionosphere (e.g., by radio remote sensing). More specifically, the influence on the observable signatures are studied, and the interaction between the spectrum of the scattered acoustic waves and the spectral properties of the inhomogeneities is highlighted. Applications to detection of weak natural and anthropogenic signals are discussed.

2:15–2:30 Break

2:30

4pPAa6. Use of the spectral-finite element method for infrasound propagation in a 3D heterogeneous environment. Katrina Burch (USACE-ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, katrina.m.burch@erdc.dren.mil), Michelle E. Swearingen (USACE-ERDC-CERL, Champaign, IL), and Ross E. Alter (USACE-ERDC-CRREL, Hanover, NH)

The use of the spectral element method (SEM), as implemented in the open-source software SpecFEM3D, is explored for the application of longitudinal wave propagation. Infrasound, $<$20 Hz, propagation using local atmospheric data and a numerical weather forecast model through a heterogeneous environment is examined. The spectral-finite element method simulates acoustic and seismic waves by particle displacement in the earth and velocity potential through the atmosphere. This method also allows for inclusion of air-to-ground coupling plus realistic topography. Modifications to the code that allow for a moving atmosphere are described. Models are designed at local distances no greater than 15 km. Furthermore, high density/velocity regions for lateral heterogeneities are implemented. Absorbing boundary conditions are applied to each of the model's sides. Simulation results are compared to real-world data collected at separate test sites and discussed. This presentation will discuss the applicability of SpecFEM3D to realistic infrasound modeling.

2:45

4pPAa7. Comparison of infrasound emissions observed during a tornado with potential fluid mechanisms. Christopher Petrin and Brian R. Elbing (Mech. & Aerosp. Eng., Oklahoma State Univ., OSU-MAE, Eng. North 218, Stillwater, OK 74078, cepetri@okstate.edu)

Infrasound may be emitted by tornado-producing storms up to 2 h before tornadogenesis, and due to their low atmospheric attenuation, these low frequencies may be detected several hundreds of kilometers away. Therefore, passive infrasound monitoring shows potential for the study and prediction of tornadoes, provided received infrasound signals can be correlated with the flow-fields of tornadoes. Literature indicates that tornadoes do cause high infrasound levels between 0.5 Hz and 10 Hz, but the radially vibrating vortex mechanism commonly proposed to explain this production [Abdullah, *Mon. Weather Rev.* **94**, 213–220] has been shown to be non-physical. Schecter [*Mon. Weather Rev.* **140**, 2080–2089] showed these limitations and showed using numerical experiments of a tornado-like vortex produced infrasound from around the storm's melting level. As this level contains diabatic processes involving hail, it also appears that hail production could be connected to tornado infrasound. In the current work, observations of atmospheric infrasound during a small tornado that occurred on 11 May 2017, will be evaluated in light of these previous studies.

3:00

4pPAa8. Exploring the use of exploding oxy-acetylene balloons for field-scale infrasound. Tracianne B. Neilsen (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Sean Maher (Univ. of California, Santa Barbara, San Diego, CA), Eric J. Lysenko, Julio A. Escobedo, Sarah A. Shaw, Margaret G. McKay, Menley R. Hawkes, Christian A. Lopez, Carla Butts (Brigham Young Univ., Provo, UT), and Robin S. Matoza (Univ. of California, Santa Barbara, Santa Barbara, CA)

Fundamental aspects of volcano-acoustic wave propagation and source effects, such as seismo-acoustic coupling, can be explored with field-scale experiments. Buried explosives are often used for these tests but require special permission and personnel. As an alternative, we explored using an environmentally friendly source of high-amplitude noise: balloons filled with a stoichiometric oxy-acetylene mix detonated with an electric match. A field-scale experiment was conducted to test the efficacy of exploding oxy-acetylene balloons to generate infrasound and modest seismic vibrations. In prior studies with these exploding balloons, the balloons were positioned above the ground to test nonlinear propagation theory [Young *et al.*, *J. Acoust.*

Soc. Am. **138**, EL305–EL310 (2015)] and quantify Mach Stem formation [Leete *et al.*, *J. Acoust. Soc. Am.* **138**, EL522–EL527 (2015)]. In the present experiment, the balloons were placed on the ground in "craters." Specifically, 17 in. oxy-acetylene balloons were exploded in three different "crater" shapes and while sitting on the ground as a control. The explosions were recorded at 100-160 m on colocated infrasound sensors, and broadband seismometers microphones at different heights. The 17 in. oxy-acetylene balloons produce significant infrasound with peak levels at 100 m of 5-13 Pa depending on type of "crater."

3:15

4pPAa9. Overview of ongoing infrasound research at the Georgia Institute of Technology. Alessio Medda, Krish Ahuja, Rob Funk, David Alvord, Darryl Dickey (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, alessio.medda@gtri.gatech.edu), Elliot Dowd (Georgia Tech Res. Inst., Smyrna, Georgia), and John Trostel (Georgia Tech Res. Inst., Smyrna, GA)

This presentation discusses ongoing research in Infrasound technologies at the Georgia Institute of Technology, Georgia Tech Research Institute (GTRI), the applied research arm of the Georgia Tech. In particular, results of a study that compared a number of commercially available infrasound sensors with several windscreen technologies are presented. Among them, comparisons obtained with a wind screen loaned to GTRI by NASA Langley and described by Ahuja and Shams in the 2017 Infrasound Workshop are also presented. Sources producing controlled infrasound under study at GTRI are also discussed. These include a sonic boom simulator, a propane vapor burner, oscillating jets, a nitrogen cannon and a low frequency acoustic driver. In addition, signatures from people moving through doorways are also presented. Each source was most effective in a given frequency range. Controlled infrasound at 0.1 Hz was obtained by several sources, among which a flame and a cold plume modulated at nominal frequencies of 0.1 Hz. Moreover, preliminary results of successful attempts at characterizing the infrasound sources and removing wind noise via wavelet analysis are also presented.

THURSDAY AFTERNOON, 16 MAY 2019 MCCREARY, 3:15 P.M. TO 5:30 P.M.

Session 4pPAb

Physical Acoustics: General Topics in Physical Acoustics I

Sam Wallen, Chair
The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

3:15

4pPAb1. Comparison of one-way and full-wave linear propagation models in inhomogeneous medium. Petr V. Yuldashev, Pavel B. Rosnitskiy (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Vera A. Khokhlova, and Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State University, Moscow 119991, Russia, and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., oa.sapozhnikov@physics.msu.ru)

One-way wave propagation models based on the parabolic approximation or its wide-angle extensions are often used for describing bounded acoustic beams. Such models are highly demanded when solving nonlinear problems that are computationally intensive and thus technically difficult to solve using full-wave approaches. When describing the propagation of a wave beam in a homogeneous medium, the one-way assumption is fulfilled exactly, and therefore, the inaccuracy is caused only by the limitations of the parabolic approximation. Such an error is significantly reduced within the wide-angle approach and completely disappears when using the exact propagator in the framework of the angular spectrum method. The situation is less obvious in the case of a heterogeneous environment, when a part of the wave energy is inevitably reflected becoming a counter-propagating wave and thus is not taken into account in the one-way approximation. The degree this phenomenon affects the accuracy of the one-way approach is still under discussion. In the current paper, a one-way propagator based on the pseudo-differential wide-angle equation is proposed. The propagator is tested for the homogeneous medium and for several configurations of media with regular and random inhomogeneities. The corresponding solutions are compared with those obtained using the k-Wave toolbox. Results of comparison show how the one-way propagator accuracy depends on the contrast and smoothness of the inhomogeneities. [Work supported by RSF No. 18-72-00196.]

3:30

4pPAb2. Acoustic wave propagation in a toroidal waveguide carrying a mean flow. Charles Thompson, Sarah Kamal, Zaineb Abdulmagid, Eyobel Haile, Samusha Najjuuko, and Carlos Araujo (ECE, UMASS, UMASS Lowell, CACT fa203, Lowell, MA 01854, sarah_kamal@student.uml.edu)

This paper describes the analysis of acoustic wave propagation in an attenuating toroidal waveguide carrying a circumferentially directed mean flow. The disposition of the standing pressure waves driven into resonance by time harmonic excitation is of particular interest. The relationship between the mean flow velocity amplitude and location of peak pressure response relative to the source position is evaluated. The conditions required for non-reciprocal scattering behavior is given.

3:45

4pPAb3. Diffuse ultrasonic transport in an unconsolidated glass bead pack. Richard Weaver and John Y. Yoritomo (Dept. of Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL, r-weaver@uiuc.edu)

We study the transport of diffuse ultrasound with frequencies of hundreds of kHz through random aggregates of d = 3.0 and 1.0 mm diameter spherical glass beads in air under static loads of 100 to 300 kPa. Highdensity polystyrene foam on top and bottom transmits the static loads while maintaining ultrasonic isolation. A floating polystyrene foam wall helps establish a uniform hydrostatic load through the 10 to 70 mm depths. Findings include a band gap extending—for the 3 mm beads—from a lower edge at about 200 kHz (that scales weakly with load and inversely with bead diameter.) Amongst the 3mm beads, we observe an upper edge to the band gap at about 900 kHz corresponding to an optical branch passband associated with the lowest internal resonance of an isolated bead. Higher optical branches are observed also. The lower edge at 200 kHz corresponds well with estimates of the upper band edge for the rotational-wave vibrations of a hexagonal close packed array of beads in Hertzian contact. The observed first arrival times correspond well with Hertzian predictions for low frequency effective longitudinal wavespeeds. Within the low frequency pass band we see diffuse transport, with diffusivities comparable to simple theoretical expectations.

4:00

4pPAb4. Impacted waves in granular media: A laboratory scale asteroids experiment. Thomas Gallot, Gonzalo Tancredi, and Alejandro Ginares (Instituto de Física, Facultad de Ciencias, Universidad de la República, Igua 4225, Montevideo 11400, Uruguay, tgallot@fisica.edu.uy)

Asteroids and small bodies of the Solar System can be considered as agglomerates of irregular boulders, therefore cataloged as granular media. Ejections of particles and dust, resulting in a cometary-type plume, can result from impacts on their surface generating waves within these bodies and potentially causing modifications in the rocks distribution. Since no asteroid seismicity data are available, we propose a laboratory scale experiment of impact-induced seismic waves in granular media. Our study focuses on the influence of static compression mimicking pressure variations induced by self-gravity on the asteroid interior. A cubic box (50 x50 x50 cm) filled with different natural and artificial granular matter is impacted with low velocity projectiles (40 to 200 m/s). An array of accelerometers records the resulting wavefield while the box is compressed to understand its dependence with the monitored internal pressure. This study is relevant to understand how asteroids reacts to kinetic energy, as is will be tested at real scale during the Asteroid Impact and Deflection Assessment mission (2022).

4:15

4pPAb5. A comparison of optical and acoustical resonances: The bisphere telescope. Cleon E. Dean and Maxim Durach (Phys. and Astronomy, Georgia Southern Univ., P.O. Box 8031, Math/Phys. Bldg., Statesboro, GA 30461-8031, cdean@georgiasouthern.edu)

A previous presentation compared acoustical and optical resonances of a Mie regime double sphere system that focused on a side scattering phenomenon that roughly mimicked a mirror [C. E. Dean and R. M. Hodges, *JASA*, **143**, 1844 (2018)]. If one thinks of these scatterers as lenses the presence of a photonic or phononic "jet" suggests a caustic region with a concentration of energy near the tip of the jet, a point analogous to the focus of a lens. Since both light and sound are reversible, there are two foci on either side of such a scatterer, arrranged symmetrically about each scatterer on the axis of the line between the centers of the two sphere system. The current research examines the case when two variable sized Mie regime scatterers are arranged so as to have the backward focus of a second scatterer on or near the forward focus of the first scatterer. This is effectively a Mie regime double sphere "telescope." Changes to far field scattering in and around the forward scattering direction are examined. This talk attempts to answer these and other questions through the use of theoretical computational acoustics models.

4:30

4pPAb6. An outdoor sound propagation model in concert with geographic information system software. Nathan D. Tipton and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 43040, ntt8@psu.edu)

As industrial technology advances, man-made noise has increasingly contributed to natural environment soundscapes. To predict how this anthropogenic noise can affect these natural environments, engineers build acoustic models over given terrain; however many current models are not compatible with common Geographic Information System (GIS) software, could become outdated due to software version updates, or are written as proprietary packages unavailable to park management. The goal of this study was to create a true open source outdoor sound propagation model compatible with (but not dependent on) outside GIS software. The model was developed to include uneven terrain, atmospheric absorption, screening, wind effects, and ground effects using ISO 9613-2, an international standard for attenuation of sound during propagation outdoors. Given sound source inputs and locations over an input Digital Elevation Map, GIS compatible file types of spatially explicit sound pressure level predictions can be produced by this model. This tool allows for ecologists and park managers to get a better understanding of how anthropogenic noise is affecting soundscapes in natural environments. [Work supported by United Technologies Corporation Professorship.]

4:45

4pPAb7. Effects of perturbing a reference atmosphere on sonic boom propagation and metrics. Lucas Wade and Victor Sparrow (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, law591@psu.edu)

There is substantial interest in the accurate noise prediction that ranges from sonic boom noise from conventional supersonic aircraft to low-boom noise (a sonic thump sound) from future aircraft designed for quiet flight. A carefully designed reference atmosphere was developed for comparing multiple sonic boom propagation programs. In the current work, that reference atmosphere was perturbed in a number of ways to assess the importance of each perturbation. The code PCBoom was used for this study, and the perturbations were for the temperature, humidity, and wind profiles. One result is that in the absence of winds, perturbing the temperature profile does not substantially affect the metrics on the ground, but perturbing the humidity profile does. [Work supported by the U.S. Federal Aviation Administration Office of Environment and Energy through ASCENT, the FAA Center of Excellence for Alternative Jet Fuels and the Environment, Project 41 through FAA Award No. 13-C-AJFE-PSU under the supervision of Sandy Liu. Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

5:00

4pPAb8. Spatio-temporal observations of temperature and wind velocity using drone-based acoustic atmospheric tomography. Anthony Finn, Kevin Rogers, Joshua Meade, Jarrod Skinner, and Amir Zhargarian (Defence & Systems Inst., Univ. of South Australia, Bldg. W, Mawson Lakes, Mawson Lakes, SA 5095, Australia, anthony.finn@unisa.edu.au)

This paper reports on a sequence of trials in which the acoustic signature of a small remotely piloted vehicle (drone) has been used to obtain spatio-temporal estimates of atmospheric temperature and wind vectors. Sound fields are recorded onboard the aircraft and by microphones on the ground. Observations are compared and the resulting propagation delays computed for each intersecting ray path transecting the intervening atmosphere. A linear model of sound speed corresponds to virtual temperature and wind velocity, plus tomographic inversion combined with regularisation, then allows vertical cross-sections and volumes of temperature and wind profile to be computed. These two- and three-dimensional profiles are represented as a lattice of elliptical radial basis functions, which enables the medium to be visualised at high levels of resolution. The technique has been used to provide spatio-temporal visualisation of atmospheric dynamics up to altitudes of 1200 m over baselines of 600 m. Independent measurements taken by co-located instruments such as a Doppler SODAR, ZephIR 300 LIDAR

and temperature sensors carried onboard drones flying within the remotely sensed atmosphere show real world performance suggests accuracies of around 0.3 °C, 0.5 m/s and 0.2 m/s for temperature, horizontal and vertical wind speeds respectively may be anticipated. The real world performance also compares very favourably to error envelopes anticipated from propagation models based on large eddy simulation.

5:15

4pPAb9. Preliminary attempts to isolate ground-radiated noise from exploding balloons. Eric J. Lysenko and Traci Neilsen (Phys., Brigham Young Univ., 700 N 56 W, 232, Provo, UT 84604, eric.lysenko@yahoo.com)

Seismo-acoustic coupling occurs when seismic wave propagation creates air-borne acoustic signals. Research is ongoing to determine methods to distinguish between sound due to seismo-acoustic coupling and purely air-borne transmission. In a field experiment, we detonated 17 in. balloons filled with a stoichiometric oxy-acetylene mix placed both on and in the ground. We attempted to isolate ground-radiated waves by constructing a portable soundproof box to deaden air-borne sound wave. The box was constructed from mass-loaded vinyl, soundproofing composite board, liquid nails, and Green Glue. This design incorporated soundproofing through decoupling, absorption, and insulation techniques. Signals observed from a microphone placed in the box are compared with those obtained on microphones outside the box at various heights. The initial blast wave was not evident inside the box. However, the loudest sound measured in the box matches a subsequent portion of signals on microphones near the ground. Testing in a reverberation chamber is done to measure the frequency response in the transmission loss through the box. These results could suggest a viable technique for isolating ground-borne acoustic waves, which could be useful in experiments where calculating the coupling effect is impractical.

THURSDAY AFTERNOON, 16 MAY 2019 CARROLL FORD, 1:15 P.M. TO 5:20 P.M.

Session 4pPP

Psychological and Physiological Acoustics and Speech Communication: Perceptual Consequences of Hearing Loss Across the Lifespan: From Children to Adults (Physiology Meets Perception)

Antje Ihlefeld, Cochair
Biomedical Engineering, New Jersey Institute of Technology, 323 Martin Luther King Blvd., Fenster Hall, Room 645, Newark, NJ 07102

Anna C. Diedesch, Cochair
Communication Sciences & Disorders, Western Washington University, Communication Sciences & Disorders, 516 High St., MS 9171, Bellingham, WA 98225

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Dept. Information Technology, Ghent Univ., Technologiepark 15, Zwijnaarde 9052, Belgium

Amanda Lauer, Cochair
Otolaryngology-HNS, Johns Hopkins University School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205

Chair's Introduction—1:15

Invited Papers

1:20

4pPP1. Factors affecting speech-in-noise and speech-in-speech recognition for school-age children with hearing loss. Lori Leibold, Jenna Browning, Ryan W. McCreery (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68124, lori.leibold@boystown.org), and Emily Buss (Univ. of North Carolina, Chapel HIll, NC)

Children with hearing loss require a more advantageous SNR than children with normal hearing to achieve comparable masked speech recognition performance. Although children with hearing loss continue to have difficulties even when appropriately fitted with hearing aids, individual differences are often substantial. This study evaluated the influence of age, degree of hearing loss, aided audibility, and daily hearing aid use on speech-in-noise and speech-in-speech recognition on a sample of 47 children (5-17 years) with bilateral, sensorineural hearing loss. Age-matched children with normal hearing were also tested. Each child completed open-set monosyllabic word recognition in two masker conditions; speech-shaped noise and two-talker speech. Thresholds for 50% correct were measured in

each masker using an adaptive tracking procedure. Compared to children with normal hearing, thresholds for children with hearing loss were elevated by an average of 7.4 dB in the noise masker and 6.5 dB in the speech masker. Preliminary results indicate that both age and aided audibility were significant predictors of performance for children with hearing loss in both masker conditions. Hearing aid use was a significant predictor of performance in noise. Degree of hearing loss was not associated with performance in either masker when aided audibility was taken into account.

1:35

4pPP2. Developmental hearing loss in conjunction with early life stress: Perceptual deficits and central auditory correlates. Merri Rosen, Yi Ye, David B. Green, and Michelle M. Mattingly (Anatomy and Neurobiology, Northeast Ohio Medical Univ., 4209 State Rte. 44, P.O. Box 95, Rootstown, OH 44272, mrosen@neomed.edu)

In children with otitis media, the conductive hearing loss (CHL) accompanying infection is a risk factor for later problems with speech perception. These perceptual deficits can persist long after auditory thresholds return to normal, suggesting they may be mediated by changes within the central auditory system. Using animal models of developmental CHL, we have demonstrated perceptual deficits for several temporally-varying signals that comprise speech. Furthermore, these perceptual deficits are correlated with impaired encoding in auditory cortex, indicating that central changes emerge from early auditory deprivation. In our transient developmental CHL model, which mimics the intermittent bouts of hearing loss experienced by children with otitis media, deficits are much alleviated by adulthood. However, early-life stress (ELS) has been described as an additional risk factor for speech problems arising from otitis media. Our data indicate that ELS alone induces deficits in the perception of temporally varying signals. Furthermore, animals experiencing both early transient CHL and ELS have perceptual deficits lasting into adulthood, the magnitude of which is greater than the sum of the individual deficits. These results raise the possibility that early life stress, alone or in conjunction with early CHL, may adverely impact speech perception in humans.

1:50

4pPP3. Consequences of auditory experience and cochlear implant stimulation on tuning and other measures obtained in pre-lingually deaf children and postlingually deaf adults. Julie G. Arenberg (Otolaryngol., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114, jbierer@uw.edu), Kelly N. Jahn (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Lindsay A. Devries (Univ. of Maryland, College Park, MD), and Mishaela DiNino (Carnegie Mellon Univ., Pittsburgh, PA)

Children and adults with moderate to severe hearing loss may obtain cochlear implants (CIs) to restore auditory perception, but auditory development differs among them. We have compared various peripheral measures across CI listeners differing in etiology, age of onset and duration of hearing loss, and duration of CI use. Several measures assess the efficacy with which CI electrodes activate their target auditory neurons in individuals with varying hearing demographics. In addition to peripheral contributions to auditory perception, central reorganization might occur when the auditory nerve is stimulated with coarse, electrical input from CIs. Evidence from neurophysiological studies in cats suggest that the central representation of spectral/spatial resolution is altered by chronic CI stimulation. In humans, psychophysical tuning curves might reflect both the spread of electrical current in the cochlea and the central representation of electrical stimuli. Understanding how chronic, electrical stimulation during auditory system development affects spectral resolution may be useful for optimizing CI programming in children and adults.

2:05

4pPP4. Neural correlates of sound-learning experiences in the auditory system: Translational candidates for hearing rehabilitation. Kasia M. Bieszczad (Psych., Rutgers The State Univ. of New Jersey, 152 Frelinghuysen Rd., Psych. Bldg. 224, Piscataway, NJ 08854, kasia.bie@rutgers.edu)

A major disconnect between traditional auditory perception research and recent neuroscience is the high propensity in the auditory system for neuroplasticity. Altered processing of reward-associated sound stimuli can contribute to adaptive behavior, such as hearing, listening, and attending appropriately to sound cues. I will the present work from animal models of learning-induced neuroplasticity in the cortical and subcortical auditory system. The data show how receptive fields and tonotopic maps in primary auditory cortex (A1) as well as the auditory brainstem response (ABR) can change when adult animals trained by pairing a tone with the availability to obtain reward alters sound coding in the auditory system. Over the course of conditioning, increases and reductions, respectively, in ABR amplitude and peak latencies predict how well animals can pick out the learned sound-frequency acoustic cue from other frequencies following conditioning. Furthermore, receptive fields in A1 have narrower tuning for a remembered sound frequency—and only in animals who successfully remember that frequency over others assessed by behavioral test. Therefore, learned sounds are preferentially processed over novel and distractor sounds following conditioning. Significant behavioral preferences for learned tones may be due, in part, to the observed changes in auditory processing across the auditory system.

2:20–2:35 Break

2:35

4pPP5. Development of language, cognition and spatial hearing abilities in children with bilateral cochlear implants. Sara Misurelli (Dept. of Surgery, Div. of Otolaryngol., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, misurelli@surgery.wisc.edu), McKenzie Klein (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI), Christi Hess (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI), and Ruth Litovsky (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI)

Cochlear implants (CIs) provide access to sound and the opportunity to communicate through spoken language for individuals who are deaf. Children who receive CIs most often have had little to no acoustic hearing; they have developed, learned, and communicated only via electrical auditory stimulation. In contrast, the majority of adults who receive CIs have had some access to acoustic hearing, either through normal hearing or through amplification via hearing aids. Our work investigates the development of language, cognition, and spatial hearing abilities in children who have bilateral CIs (BiCIs), many from an early age. The following will be discussed: (1) factors that may predict better and faster language development and cognition in children with CIs and (2) development of the ability to use cues for spatial hearing with degraded electrical signals. Our results thus far indicate that children with BiCIs are able to develop the ability to use spatial cues to aid in source segregation, particularly if they receive BiCIs at an early age. In addition, preliminary results reveal that maternal education and IQ may predict development of improved language scores over time for children with BiCIs.

Contributed Paper

2:50

4pPP6. Neural sensitivity to dynamic binaural cues: Human electroencephalogram and chinchilla single-unit responses. Ravinderjit Singh, Hari M. Bharadwaj, and Mark Sayles (Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, singh415@purdue.edu)

Animals encounter dynamic binaural timing information in broadband sounds such as speech and background noise due to moving sound sources, self motion, or reverberation. Most physiological studies of interaural time delay (ITD) or interaural correlation (IAC) sensitivity have used static stimuli; neural sensitivity to *dynamic* ITD and IAC is rarely systematically addressed. We used a system-identification approach using maximum-length sequences (MLS) to characterize neural responses to dynamically changing ITDs and IACs in broadband sounds. Responses were recorded from humans (electroencephalogram; EEG) and from single neurons in terminally anesthetized chinchillas (auditory nerve fibers; ANFs). Chinchilla medial superior olive (MSO) responses were simulated based on binaural coincidence from recorded ANF spike times in response to left- and right-channel input. Estimated ITD and IAC transfer functions were low-pass, with corner frequencies in the range of hundreds of Hz. Human EEG-based transfer functions, likely reflecting cortical responses, were also low-pass, but with much lower corner frequencies in the region of tens of Hz. Human behavioral *detection* of dynamic IAC extended beyond 100 Hz consistent with the higher brainstem limits. On the other hand, binaural unmasking effects were only evident for low-frequency ITD/IAC dynamics in the masking noise.

Invited Paper

3:05

4pPP7. Visual influences on auditory spatial processing. Yi Zhou (College of Health Solutions, Arizona State Univ., 975 S. Myrtle Ave., Coor 3470, Tempe, AZ 85287, yizhou@asu.edu)

Sensory experience is the result of a multisensory analysis of the environment around us. When information is properly integrated, visual cues facilitate auditory localization. To investigate the spatial and temporal rules of contingency in multisensory integration, a majority of previous studies have focused on sensory space within the field of vision. But the spaces encoded by vision and audition do not always align with each other. For foveal species such as humans and monkeys, the visual field is restricted to frontal space, whereas the auditory field is panoramic, covering the entire frontal and rear space. The rear sensitivity provided by spatial hearing is critical for avoiding unseen dangers coming from behind. The rear space, however, has been largely overlooked in multisensory research. In this talk, I will present recent work related to vision's role in panoramic spatial hearing in humans, the changes in visual bias observed in human listeners with chronic unilateral hearing loss, and findings concerning visual modulation of spatial responses of single neurons in the marmoset auditory cortex. Based on these results, I will discuss the challenges of implementing existing theories of multisensory spatial perception in neural circuits.

Contributed Paper

3:20

4pPP8. Cortical reorganization following auditory spatial training in listeners with sensorineural hearing impairment: A high-density electroencephalography study. K. V. Nisha and Ajith U. Kumar (Dept. of Audiol., All India Inst. of Speech and Hearing, Manasagangothri, Mysore, Karnataka 570006, India, nishakv1989@gmail.com
)

The present study is intervention-based research aimed at remediation of spatial deficits in listeners with sensorineural hearing impairment (SNHI), through the use of virtual acoustic technology. A mixed group design comprising both within (pre-test, post-test control group design) and across the groups (standard group) comparisons were performed. The study included 37 participants, who were divided into three groups. Groups I and II consisted of SNHI listeners, while group III comprised normal hearing (NH) listeners. The study was conducted in three phases. At the pre-training phase, electroencephalographic (EEG) recordings were acquired from all the three groups using spatial deviants presented in P300 paradigm. Following this, group I listeners underwent virtual acoustic space training (VAST), and post-training EEG recordings were obtained. EEG recordings were also acquired from group II listeners in second evaluation without subjecting them to any formal spatial training. Results of unpaired t-tests, grand average waveforms and scalp topographies of offline processed waveforms revealed significant differences between SNHI and NH listeners. Furthermore, spatio-temporal analyses showed the emergence of new scalp maps in post-training phase in trained listeners and no topographic changes in untrained SNHI group, suggestive of benefit derived from VAST right at the fundamental level (cortical) of spatial processing.

Invited Paper

3:35

4pPP9. Speech-in-noise recognition examined in individuals with normal hearing sensitivity and tinnitus using behavioral and brain imaging methods. Fatima T. Husain (Beckman Inst. of Adv. Sci. and Technol., Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, husainf@illinois.edu) and Yihsin Tai (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Although we know tinnitus can cause concentration problems, its contribution to communication difficulties has not been well-studied. In a series of experiments, we have investigated Speech-in-noise (SiN) performance, in those with tinnitus and normal hearing sensitivity using a variety of methods, including behavioral, otoacoustic emissions, structural MRI and in a proposed study, ERP/ EEG. We found that tinnitus patients with normal hearing sensitivity did not have a general speech-in-noise deficit (Tai and Husain, *JARO*, 2018). Instead, our findings indicated that the tinnitus group performed significantly worse only under the 5-dB signal-to-noise ratio (SNR) condition. Additionally, the SiN performance in tinnitus patients was found to be significantly correlated with the perceptual factors related to tinnitus, such as perceived loudness, and was worse in the left ear. We are currently investigating (1) how the left ear appears to be more affected in tinnitus by using structural MRI and (2) whether there is any correlation between tinnitus pitch and consonant recognition. For both the latter studies, we are also contrasting the normal hearing tinnitus group with a hearing loss tinnitus group and other control groups.

Contributed Paper

3:50

4pPP10. Speech auditory brainstem responses in adult hearing aid users: Effects of aiding and background noise, and prediction of behavioral measures. Karolina Kluk, Ghada BinKhamis (Manchester Ctr. for Audiol. and Deafness (ManCAD), The Univ. of Manchester, Oxford Rd., Manchester m13 9pl, United Kingdom, karolina.kluk@manchester.ac.uk), Antonio Elia Forte, Tobias Reichenbach (Dept. of BioEng., Ctr. for Neuro-Technol., Imperial College London, London, United Kingdom), and Martin O'Driscoll (Manchester Ctr. for Audiol. and Deafness (ManCAD), The Univ. of Manchester, Manchester, United Kingdom)

The aim of the study was to investigate the effect of aiding (hearing aids) and background noise on Auditory Brainstem Responses to short consonant vowel speech (Speech-ABRs), and to assess the predictive value of these responses in adults with a bilateral sensorineural hearing loss. Speech-ABRs evoked by a 40-ms [da] were recorded from 98 adult hearing-aid users via loudspeaker stimulus presentation with and without a hearing aid, in quiet and in 2-talker babble using a two-channel vertical electrode montage. Behavioral speech perception in noise and/or aided self-reported speech understanding were assessed. Aided speech-ABRs had earlier peak latencies, larger peak amplitudes, and larger F0 encoding amplitudes compared to unaided speech-ABRs. Background noise resulted in later F0 encoding latencies but did not have an effect on peak latencies and amplitudes, or on F0 encoding amplitudes. Speech-ABRs were not a significant predictor of any of the behavioral or self-report measures. Speech-ABRs are not a good predictor measure of speech-in-noise performance or self-reported speech understanding with hearing aids. However, they may have potential for clinical application as an objective measure of speech detection with hearing aids. [Work supported by EPSRC EP/M026728/1, Saudi Arabian Ministry of Education, NIHR MBRC.]

Invited Papers

4:05

4pPP11. Characterizing noise-induced hidden hearing loss in veterans. Naomi Bramhall (VA RR&D NCRAR, 3710 SW US Veterans Hospital Rd., P5 NCRAR, Portland, OR 97239, naomi.bramhall@va.gov)

Cochlear synaptopathy, the partial loss of auditory nerve synapses onto inner hair cells, has been proposed as a possible source of hyperacusis, some forms of tinnitus, and difficulty understanding speech in background noise. In animal models, cochlear synaptopathy is associated with a reduction in the amplitude of wave I of the auditory brainstem response (ABR) and can occur even when auditory thresholds are normal. This presentation will discuss noise exposure-related changes to several auditory physiological measures, including the ABR, in young military Veterans with clinically normal pure tone thresholds. Veterans show differences from non-Veteran controls even after statistically adjusting for group differences in sex and otoacoustic emissions, suggestive of synaptic or neuronal loss. While these physiological changes do not appear to be associated with decreased performance on standard speech-in-noise tests, they are associated with the report of frequent or constant tinnitus. Although post-mortem histological analysis would be necessary for confirmation, these data are consistent with animal models of cochlear synaptopathy and suggest that synaptopathy or "hidden hearing loss" may occur in response to high intensity noise exposure in humans and be correlated with tinnitus.

4:20–4:35 Break

4:35

4pPP12. Using perceptually relevant speech stimuli for making physiological measurements of the human auditory brainstem. Ross K. Maddox (Biomedical Eng. and Neurosci., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rm. 5.7425, Rochester, NY 14642, ross.maddox@rochester.edu)

Speech perception is one of the most important functions of the auditory system. The brainstem is an essential part of this process. While studies of natural speech processing typically employ behavioral tasks, and more recently cortical electroencephalography, studies of the human brainstem have been limited (by necessity) to short stimuli like clicks, tonebursts, and single syllables. We recently described a method for presenting continuous naturally uttered speech and deriving the auditory brainstem response. This method makes it possible to create engaging tasks using natural speech while making simultaneous physiological measurements, with applications to a wide range of scientific questions. One such question, with a history of mixed findings, is that of selective attention's role in brainstem processing. We will discuss our work using the speech-derived auditory brainstem response in a two-talker listening task. In keeping with history, our results seem to differ from those of other recent studies using a similar technique. We will also discuss preliminary work at adapting the technique for audiological purposes, in hopes that using speech stimuli will provide a more accurate clinical predictor of speech perception.

4:50

4pPP13. Aging and hearing loss effects on neural speech processing. Samira B. Anderson (Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., 0100 Lefrak Hall, College Park, MD 20742, sander22@umd.edu)

The effects of age-related hearing loss are pervasive, resulting in declines in social, emotional, and cognitive function. As the world becomes a quieter place, the decrease in sensory input from the auditory periphery may trigger homeostatic mechanisms to preserve a stable rate of neural firing at higher levels of the auditory system from brainstem to cortex. For example, decreased inhibitory neurotransmission increases neural excitability, preserving the sensation of loudness for moderate to moderately loud conversational speech levels. However, this change in the balance of excitatory and inhibitory neurotransmission may disrupt the brain's ability to follow the rapid acoustic changes that are characteristic of running speech. Age-related disruptions in auditory processing of synthesized syllables and naturally-produced words and sentences have been demonstrated using electrophysiology (EEG) and magnetoencephalography (MEG). This presentation will review a series of EEG and MEG studies demonstrating effects of aging and/or hearing loss that vary depending on factors associated with type of hearing loss, stimulus choice, and primary neural source (midbrain versus cortex). Clinical implications for hearing loss management will be discussed. [Work supported by NIH-NIDCD, R21 DC015843-01.]

Contributed Paper

5:05

4pPP14. Nano, micro, and macro-scale effects on cochlear tuning. Aritra Sasmal and Karl Grosh (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, asasmal@umich.edu)

Reconciling the highly tuned and nonlinear basilar membrane (BM) response at the base with the nearly low-pass and weakly nonlinear response at the apex has presented a longstanding challenge to cochlear mechanics modelers. Recent experiments have shown that the BM centric view of cochlear mechanics is incomplete and have highlighted the importance of modeling and measuring the dynamics of the organ of Corti (OoC). Here, we describe a new computational model of the guinea pig cochlea that can correctly simulate the response at all frequencies. The model shows that the electromotile force from the outer hair cells modulate the differential motion between the reticular lamina and the BM. Model calculations at the apex show that the geometric taper of the scalae duct as well as the cytoarchitecture of the OoC breaks the scaling symmetry observed at the base. Further, the model predicts that the neural tuning at the base is primarily governed by the macroscopic dynamics of the cochlear partition, while the micro-scale fluid dynamics and the nano-scale channel dynamics dominate the neural tuning at the apex. Overall, the model provides a physiological explanation for the differences between high and low frequency hearing observed in psychophysical experiments. [Work supported by NIH-R01-04084.]

THURSDAY AFTERNOON, 16 MAY 2019 GRAND BALLROOM C, 1:30 P.M. TO 5:30 P.M.

Session 4pSC

Speech Communication: Perception (Poster Session)

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

4pSC1. Auditory sensation of impulsivity and tonality in vocal fry. Philipp Aichinger, Imme Roesner (Dept. of Otorhinolaryngology, Div. of Phoniatrics-LogopeDC, Medical Univ. of Vienna,Waehringer Guertel 18-20, Univ. Klinik HNO, Vienna 1090, Austria, philipp.aichinger@meduniwien.ac.at), and Jean Schoentgen (Dept. of Bio-, Electro- and Mech. Systems, Faculty of Appl. Sci., Université libre de Bruxelles, Brussels, Belgium)

Vocal fry is a phonation type characterized by nearly complete vocal tract damping during the closed glottal cycle phase, caused by a low vocal frequency in combination with a long glottal closed phase. Auditorily, vocal fry is characterized by the sensation of individual glottal cycles. Vocal fry may be (para-)linguistically relevant, but it may also be a symptom of a voice disorder. The aim of the study is to develop predictive models of the presence of vocal fry based on data from auditory experimentation using synthetic stimuli. Predictors are the vocal frequency, the glottal open quotient, and the glottal pulse skewness. The vocal tract is kept constant. Tests are conducted with stimuli that are temporally homogeneous in terms of voice quality, as well as with stimuli that contain neutral-fry-neutral voice quality transitions. Listeners rate tonality, impulsivity, and naturalness of stimuli on 7-point scales, as well as the presence of vocal fry on a dichotomous scale. Results show that perceived vocal fry is correlated with an increase in perceived impulsivity and a decrease in perceived tonality of the voice. The most important predictor of vocal fry is vocal frequency, whereas open quotient and skewness appear to play a minor role.

4pSC2. Effects of talker variability on categorization of spectrally degraded vowels. Emily Dickey and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., University of Louisville, Louisville, KY 40292, eadick01@louisville.edu)

When spectral properties differ between earlier (context) and later (target) sounds, categorization of later sounds becomes biased through *spectral contrast effects* (SCEs). Recent work has shown that talker variability diminishes SCEs: shifts in vowel categorization were smaller when context sentences were spoken by 200 talkers than one talker [Assgari and Stilp, *JASA* (2015)]. CI users' speech categorization is also influenced by SCEs [Feng and Oxenham, *JASA* (2018)] but are known to struggle with talker discrimination. Here, we tested whether talker variability affected context effects in spectrally degraded speech perception. Listeners categorized target vowels varying from "ih" as in "bit" to "eh" as in "bet" following 200 context sentences spoken by one or 200 talkers (from Assgari and Stilp, 2015). Sentences had 5-dB spectral peaks added to low-F1 (100–400 Hz) or high-F1 (550–850 Hz) frequencies (to produce SCEs) then were noise vocoded at different spectral resolutions. At 4 and 8 channels, the experiment was too difficult to produce reliable results (flat categorization functions). At 12 and 24 channels, SCEs occurred but did not significantly differ across one-talker and 200-talker conditions. Talker variability does not appear to affect perception of spectrally degraded speech in the same way it does for normal-hearing listeners.

4pSC3. Mandatory dichotic integration of second-formant information: Mismatched contralateral sine bleats have predictable effects on place judgments in consonant-vowel syllables. Brian Roberts, Robert J. Summers (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk), and Peter J. Bailey (Psych., Univ. of York, York, United Kingdom)

Speech-on-speech informational masking may arise because the interferer disrupts processing of the target (e.g., capacity limitations) or corrupts it (e.g., intrusions into the target percept). The latter should produce predictable errors. Listeners identified the consonant in monaural three-formant analogues of approximant-vowel syllables, lying along a place-of-articulation continuum ([w]-[l]-[y]). There were two eleven-member continua; the vowel was either high-front or low-back. Continuum members shared F1 and F3 frequency contours; they were distinguished solely by the F2 contour prior to the steady portion. Continuum members also shared amplitude contours and fundamental frequency (130 Hz). Targets were always presented in the left ear. For each continuum, the F2 frequency and amplitude contours were also used to generate interferers with different source properties—sine-wave analogues of F2 (sine bleats) RMS-matched to their buzz-excited counterparts. Accompanying each continuum member with a matched sine bleat in the contralateral ear had little effect, but accompanying each member by a fixed mismatched bleat (1, 6, or 11) produced systematic and predictable effects on category judgments. This outcome indicates that informational masking by interferers involved corruption of target processing as a result of mandatory dichotic integration of F2 information, despite the grouping cues disfavoring this integration. [Work supported by ESRC.]

4pSC4. Influence of semantics on the perception of gender and femininity. Serena Piol and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, sp4864@nyu.edu)

Research on the perception of femininity of a speaker has either focused on acoustic parameters (e.g., F0) or on lexical differences (e.g., mauve/purple). This study simultaneously examined how acoustic parameters and semantic content affect the perception of a speaker's gender and femininity. Three speaker groups (cismen, ciswomen, and transwomen) produced sentences that were categorized as containing masculine lexical items (e.g., The boy gave the football a kick), feminine items (e.g., The little girl cuddled her doll), or neutral items (e.g., Airmail requires a special stamp). Listeners were first asked to identify the gender (male/female) and then asked to rate femininity on a visual analog scale. Results revealed no significant differences in femininity based on the lexical category of the sentences. Consistent with previous findings, average F0 predicted femininity ratings. Despite previous research showing differences in speaking rate for cismale and cisfemale speakers, our data revealed no effect of speaking rate on femininity rating.

4pSC5. The effect of altered sentence rhythm on timing judgments. Dylan V. Pearson, Yi Shen (Speech and Hearing Sci., Indiana Univ.-Bloomington, 200 S. Jordan Ave., Bloomington, IN 47401, dylpears@iu.edu), J. Devin McAuley (Dept. of Psych., Michigan State Univ., East Lansing, MI), and Gary R. Kidd (Speech and Hearing Sci., Indiana Univ.-Bloomington, Bloomington, IN)

Successful speech understanding requires the listener to accurately anticipate the temporal onsets of individual words in running speech. The present study investigated listeners' sensitivity to temporal deviations in sentences with natural or modified speech timing. Subjects listened to sentences in which a portion of speech preceding the final word was replaced by a silent gap. On each trial, an intact sentence was presented, followed by two versions of the sentence with a silent gap: one with the correct timing for the gap (i.e., equal to the duration of the missing speech) and one with altered gap timing (longer or shorter than the missing speech). Listeners judged which version had the altered timing. An adaptive procedure was used to estimate thresholds for the detection of altered timing for early-onset (shortened gap) and late-onset (lengthened gap) final words. In separate conditions, the rhythm of the sentence preceding the gap was either unaltered or rate-modulated according to a sinusoidal modulator. Results showed that the ability to identify the correct gap timing was adversely affected by the manipulation of sentence rhythm, and in both intact and altered rhythmic contexts, listeners were better at detecting early final word onsets than late onsets.

4pSC6. A database of English multisyllabic words for speech perception research. Cody Elston (Commun. Disord. & Sci., Rush Univ., 212 14th Ave., Sterling, IL 61081, codyelston@gmail.com), Brendan Prendergast, Madeleine Thomas, Mark Partain, Elizabeth Butler, Stanley Sheft, and Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ., Chicago, IL)

Recognition of individual words is frequently used to investigate speech intelligibility and underlying perceptual processing. Traditionally, the majority of such studies in English have utilized monosyllabic and, on occasion, disyllabic words and spondees. Although multisyllabic words have

been used extensively to investigate visual processing and lexical organization in reading, little research exists on the auditory perception of spoken multisyllabic English words. The present database was designed to provide materials to facilitate further research into the intelligibility and perceptual processing of spoken multisyllabic words. The database consists of five sections of 1–5 syllable words each. Individual words in each section were generated from the English Lexicon Project website. All words in the database were recorded by a male native speaker of American English, separated into individual word audio files and equalized in root-mean-square energy. Each syllable section contains 1125 words that vary in duration, frequency of occurrence and phonological neighborhood density. With a large number of words in each syllable section, shorter word lists can be selected from each section and matched on specific lexical characteristics. The database recordings are available free of charge for research purposes to improve understanding of perceptual processing of multisyllabic words.

4pSC7. The role of gender expectations on word recognition. Dylan V. Pearson and Tessa Bent (Speech and Hearing Sci., Indiana Univ.-Bloomington, 200 S. Jordan Ave., Bloomington, IN 47401, dylpears@iu.edu)

Socio-indexical and linguistic information bi-directionally interact during speech processing. Information about a speaker's age, gender, or ethnicity, conveyed through speech or visual cues, can influence how acoustic-phonetic cues are mapped to phoneme categories. For example, in McGowan (2015), Chinese-accented English sentences were presented along with a Chinese face (congruent), Caucasian face (incongruent), or no detailed visual information. Intelligibility scores were significantly higher in the congruent than the incongruent condition. Here, we investigate whether similar effects are observed for talker gender. Participants orthographically transcribed sentences mixed with noise from native American English male and female talkers. A gender congruent or incongruent visual face prime was presented before each sentence. In a control condition, different participants completed the task without the inclusion of visual face primes. Results showed that female talkers were significantly more intelligible than male talkers. Further, a gender congruency benefit was observed for female talkers, but not for male talkers. No incongruency cost was found; intelligibility scores in the incongruent and no-face control conditions did not differ. Although congruency effects were only observed with female talkers, the results suggest that expectations about speaker gender can influence word recognition accuracy similar to previously reported ethnicity effects.

4pSC8. Segmental duration as a cue to sentence structure. Sten Knutsen, Karin Stromswold, and Dave F. Kleinschmidt (Psych., Rutgers Univ., 152 Frelinghuysen Rd., Piscataway, NJ 08854, sten.knutsen@rutgers.edu)

In order to parse speech in real time, listeners should use any informative cues available. Here, we investigate the role of segmental duration. Previous work has found statistically significant differences in the mean durations of analogous segments across different lexical/syntactic structures. However, a difference in means does not necessarily mean that the distributions of these durations make individual token durations sufficiently informative to be a useful cue. The goal of this work is to use production data to quantify how informative segmental duration is about syntactic/lexical structure. Our model is based on an ideal listener model, where we assume listeners have implicit knowledge of segmental duration distributions for active and passive sentences. Given these distributions, the model can infer the posterior probability that a particular token belongs to one distribution or the other. After implementing our model in a Bayesian classifier, our results indicate there is indeed sufficient information contained in individual token durations so as to be useful in real-time sentence processing. Furthermore, we modeled listener behavior in a gating task with syntactically ambiguous sentences truncated before disambiguating morphosyntax and achieved 74% accuracy in predicting syntactic outcome, similar to accuracy reported in behavioral studies (62%–84%).

4pSC9. Seeing is believing: The role of the visual stimulus in perception of rounded vowels in Canadian French. John M. Sances (Linguist, Univ. of New Mexico, 1 University of New Mexico, MSC03 2130, Albuquerque, NM 87131, jsances@unm.edu)

For face-to-face communication, the visual stimulus has been shown to be important in speech perception. For vowels, lip protrusion of rounded vowels is the most visually salient signal. Rounding is a contrastive feature of French vowels, both front and back. Tests of native French speakers' lip-reading ability show that front rounded vowels are perceived poorly; the vowel perceived tends to be the back rounded counterpart (Tseva and Cathiard, 1990). Other work (Benoit *et al*., 1994) has found that native French speakers often perceive the auditory signal for front rounded /y/ as /i/, the unrounded version. Adding visual information to the stimulus drastically increases accuracy. Another study corroborates this in showing that rounding is the least salient perceivable feature in the auditory stimulus, but the most salient in the visual stimulus (Robert-Ribes *et al*., 1998). The current work extends these findings in a comprehensive experiment using audio, visual, and audio-visual stimuli with the two sources both matched and mismatched. As found previously, rounding was the most salient feature visually. However, rounding was also very salient in the auditory stimulus, contradicting previous research. In the audio-visual mismatched stimuli, listeners tended to favor the auditory signal over the visual signal almost exclusively.

4pSC10. Angry prosody slows responses to simple commands. Aleah D. Combs (Linguist, Univ. of Kentucky, Lexington, KY), Emma Kate Calvert (Commun. Sci. and Disord., Univ. of Kentucky, Lexington, KY), and Kevin B. McGowan (Linguist, Univ. of Kentucky, 1415 Patterson Office Tower, Lexington, KY 40506, kbmcgowan@uky.edu)

Previous research has found that emotional prosody can interact with speech perception and listeners' processing of the meaning of particular word/emotion pairings (Kim and Sumner, 2017). What remains unclear is how this interactive processing can affect behavioral responses such as responses to imperatives. To answer this question, 42 participants were presented with a series of commands read either with angry prosody, happy prosody, or neutral prosody and were instructed to press the requested button on a response box as quickly and accurately as possible. All emotional states were performed by a trained actor, rather than induced, and the stimuli were independently rated for accuracy of performance. On average, participants responded roughly 50ms slower to the commands which were performed with "angry" prosody. There was no difference between responses to "happy" and "neutral" prosody commands. This difference in response time may be due to the heightened neurological responses to angry stimuli (Frueholz and Didier, 2013). These results are consistent with a model of speech perception in which linguistic and social information are processed simultaneously and interactively (Sumner *et al*., 2014) but not with a model in which emotional aspects of the speech signal or discarded or irrelevant to perception.

4pSC11. Dialect-specific features enhance perception of phonetic imitation of unfamiliar dialects. John P. Ross, Kevin D. Lilley, Cynthia G. Clopper (Linguist, Ohio State Univ., 1712 Neil Ave, Oxley Hall 100, Columbus, OH 43210, ross.1589@osu.edu), Jennifer Pardo (Psych., Montclair State Univ., Montclair, NJ), and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Talkers reproduce speech features of their interlocutors through phonetic imitation. In this study, the effects of experience with a dialect on phonetic imitation and the perceptibility of that imitation were explored. Talkers with New York City and General American accents repeated isolated words after model talkers with New York City and General American accents in a shadowing task. Half of the target words contained phonetic features differing between the two accents, including the stressed vowel in words like

"cauldron, the stressed vowel in words like carriage, and the initial fricative in words like stranger. The other half contained no distinguishing dialect features. Participants from the Midwestern United States completed an AXB task assessing the perceptual similarity of the repeated words to the original stimuli. The results demonstrated that accuracy was above chance overall, suggesting imitation across shadower and model talker accents. Additionally, a significant interaction between the presence of dialect-specific features and shadower dialect was observed: the presence of dialect-specific features facilitated identification of imitations by New York City shadowers, but had no effect on identification of imitations by General American shadowers. These findings suggest that the perception of phonetic imitation of unfamiliar dialects is enhanced by iconic dialect features.

4pSC12. Bidirectional effects of priming in speech perception: Social-to-lexical and lexical-to-social. Dominique A. Bouavichith, Ian C. Calloway, Justin T. Craft, Tamarae Hildebrandt, Stephen J. Tobin, and Patrice S. Beddor (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, dombouav@umich.edu)

Previous perceptual research demonstrates that providing listeners with a social prime, such as information about a speaker's gender, can affect how listeners categorize an ambiguous speech sound produced by that speaker. We report the results of an experiment testing whether, in turn, providing listeners with a linguistic prime, such as which word they are about to hear, affects categorization of that speaker's gender. In an eye-tracking study testing for these bidirectional effects, participants (i) saw a visual prime (gender or lexical), (ii) heard an auditory stimulus drawn from a matrix of gender (female-to-male) and sibilant frequency (shack-to-sack) continua, and (iii) looked to images of the non-primed category. Social prime results replicate earlier findings that listeners' /s-ʃ/ boundary can shift via visual gender information. Additionally, lexical prime results indicate that listeners' judgments of speaker gender can shift with visual linguistic information. These effects are strongest for listeners at category boundaries where linguistic and social information are least prototypical. In regions of high linguistic and social prototypicality, priming effects are weakened or reversed. The results provide evidence of a bidirectional link between social and linguistic categorization in speech perception and its modulation by the stimulus prototypicality.

4pSC13. Perceptual preference for falling tones over rising tones: A study of Mandarin Chinese. Yuyu Zeng, Allard Jongman, Joan A. Sereno, and Jie Zhang (Linguist, The Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, yzengae@ku.edu)

Typological studies have shown that there are more falling tones than rising tones in tone languages, including Chinese. We test the hypothesis that this may be due to a perceptually-based advantage for falling tones over rising tones. Two acoustically comparable (and matched for naturalness) tonal continua in Mandarin (level-falling T1-T4, and level-rising T1-T2) were created. Identification and discrimination results were obtained from 14 native Mandarin speakers. The results revealed that it is easier to identify a falling tone than a rising tone; that is, listeners require a smaller F0 difference between onset and offset to distinguish a falling tone from a level tone as compared to a rising tone from a level tone. Additionally, there are several hints of better discrimination for the falling continuum. This disagrees with our and others' Mandarin production data, which show that the rising tone is closer to the level tone than the falling tone is, hence a production-perception dissociation. We propose that, historically, Chinese listeners' greater sensitivity to the level-falling contrast has resulted in the preponderance of falling tones over rising tones found across Chinese languages, and this proposed explanation may be applicable to other tone languages as well.

4pSC14. Acoustic cues to perception of labialized stops in a merger in progress. John Culnan (Dept. of Linguist, Univ. of Arizona, Tucson, AZ 85721, jmculnan@email.arizona.edu) and Suki Yiu (Linguist, Univ. of Hong Kong, Hong Kong, Hong Kong)

In Hong Kong Cantonese, the labialized and plain velar (/k^{w}/ and /k/) are undergoing a merger where both may be produced as [k] before the vowel /ɔ/. This study examines the role of acoustic cues to labialization in the perception of velars in Hong Kong Cantonese, and whether these cues are utilized differently in the merger environment than other environments. Native listeners of Hong Kong Cantonese completed an identification task comprised of LPC resynthesized words of Cantonese in the merger environment and three other (unrounded) vowel environments. As anticipated initial results suggest that F2 transition is a less important cue for velars preceding /ɔ/ than those preceding other vowels, although surprisingly F1 transition may only be important for velars preceding /ɔ/. An examination of the importance of intensity suggests that this cue is not used in distinguishing labialized from plain velars. Finally, perception accuracy results on unchanged productions suggest that distinguishing labialized velars from plain velars is a more difficult task in the merger environment than other environments. Taken together, the pilot results suggest that native listeners of Hong Kong Cantonese rely more on multiple acoustic cues in environments where a single cue provides less certain information.

4pSC15. Reading aloud in a clear speaking style may interfere with sentence recognition memory. Sandie Keerstock and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, The University of Texas at Austin, 305 E. 23rd St. CLA 4.400 E9 Mail Code: B5100, Austin, TX 78712, keerstock@utexas.edu)

Previous research has shown that native and non-native listeners' recognition memory is higher for sentences previously heard in clear speech (CS) than in conversational speech (Keerstock and Smiljanic, 2018). The present study investigated whether speaking clearly also enhances *talkers*' sentence recognition memory. The production effect (MacLeod *et al*., 2010) revealed superior retention of material read aloud relative to material read silently during an encoding phase. Vocal production that included an additional distinct element, such as speaking loudly, produced even greater memory benefits than reading aloud (Quinlan and Taylor, 2013). Production of the exaggerated articulatory and acoustic cues in CS could thus lead to enhanced sentence recognition memory relative to conversational style. Native and non-native English speakers produced alternating blocks of 10 conversational and CS sentences for a total of 60 unique sentences. In the test phase, they identified 120 sentences as old (from exposure) or new (distractors). Unexpectedly, preliminary results show lower sentence recognition memory for sentences produced in CS than in conversational speech for both groups of talkers. The results suggest that producing CS, unlike perceiving it, interferes with recognition memory. Allocating cognitive resources to producing hyper-articulated speech may limit their availability for storing information in memory.

4pSC16. Speaker-listener dialect differences and spoken word recognition: Evidence from massive auditory lexical decision. Filip Nenadic, Matthew C. Kelley, and Benjamin V. Tucker (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

The difficulty associated with perceiving an unfamiliar dialect has been shown in several studies using novel dialects, synthesized vowels, or recorded sentences (e.g., Goslin *et al*., 2012; Wright and Souza, 2012; Maye *et al*., 2007). One of the goals of the Massive Auditory Lexical Decision dataset (Tucker *et al*., 2018) is to investigate the effects of speaker-listener dialect mismatches on spoken word recognition. In three separate auditory lexical decision experiments, monolingual native speakers of English from different dialect regions (231 speakers of western Canadian English recruited in Edmonton, Alberta; 77 speakers of southwestern American English recruited in Tucson, Arizona; and 53 speakers of eastern Canadian English recruited in Halifax, Nova Scotia) each responded to a subset of the same word and pseudoword stimuli recorded by one male speaker of western Canadian English. Therefore, some of the participants had greater experience with the speaker's dialect than others, where the Edmonton participants had the most experience, the Tucson participants had the least, and the Halifax participants were in the middle. We discuss the results of the comparison of responses from these three dialect groups and their implications to speech perception and comprehension of less familiar dialects.

4pSC17. Investigating the comprehension and perception of reduced speech with pupillary response. Benjamin V. Tucker (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca), Vincent Porretta (Psych., Univ. of Windsor, Windsor, ON, Canada), and Yoichi Mukai (Linguist, Univ. of AB, Edmonton, AB, Canada)

Spontaneous, casual speech is highly variable, in part due to reduction processes. Listeners handle these reductions in everyday communication; however, these forms present challenges for models of speech perception and lexical processing. Previous research has found that reaction times to reduced word-medial stops are longer, indicating that they are more difficult to process than words with unreduced word-medial stops (Tucker, 2011). The current study examines spoken word processing (as measured by pupil dilation) of reduced and unreduced word-medial stops to determine (a) if the pupillary response to reduced forms corresponds to reaction time results and (b) when in time any differences emerge. Thirty-nine native speakers of North American English completed a listen-and-repeat task in which 80 isolated disyllabic reduced and unreduced word-medial /d/ and /g/ items (40 of each phoneme) were presented. The pupil size data and speech productions are analyzed and will be reported. The results indicate significantly greater pupil dilation for reduced /d/ and /g/. Words containing /d/ elicited greater dilation than those containing /g/, for reduced and unreduced forms. This suggests that, although word-medial stop reduction is frequent in English, an increased processing load is incurred, mirroring previous reaction time results.

4pSC18. Experienced listeners' differential perception of enhancing and non-enhancing cue pairs. Meng Yang (Linguist, Univ. of California, Los Angeles, 3125 Campbell Hall, Los Angeles, CA 90095, mengyang@ucla.edu)

The perception of a pair of enhancing cues, pitch and breathiness, and non-enhancing cues, pitch and vowel duration, were compared in a cue-shifting experiment with Hani listeners who have experience with both cue pairs in the same tense-lax contrast. Results show that, when the correlation between the non-enhancing cues was the reverse of their experience, it was equally difficult for Hani listeners to shift attention from pitch to vowel duration as from vowel duration to pitch. For the enhancing cues, difficulty with cue-shifting was asymmetric; shifting from pitch to breathiness was easy, but shifting from breathiness to pitch was difficult. This asymmetry may occur because the perceptual dependence between pitch and breathiness is unidirectional: breathiness is associated with pitch but not vice versa. Thus, listeners experience interference only when shifting attention from breathiness to pitch. The difference between two cue pairs and the direction of asymmetry found for the enhancing cue pair is consistent with earlier results from English listeners who do not have phonemic experience with either pair. Collectively, these results indicate that enhancing cues are perceived differently from non-enhancing cues, even when listeners' language experience with cue pairs is equated.

4pSC19. Segmental and suprasegmental communicative information: Speech versus emotion recognition thresholds. Shae D. Morgan (Otolaryngol. Head and Neck Surgery and Communicative Disord., Univ. of Louisville, 627 S. Preston St., Ste. 220, Louisville, KY 40292, shae.morgan@louisville.edu)

Effective communication not only depends on what is said (segmental information), but how it is said (suprasegmental information). Research has clearly demonstrated the impact that intelligible segmental information has on communication. Less studied, however, is the availability of suprasegmental information during communicative interactions. Tests of speech recognition or word discrimination are commonly used to assess segmental information in the speech signal. No similar tests have been employed to detect the threshold for detection of suprasegmental information in speech. In this study, we examined thresholds of suprasegmental information (i.e., talker emotional state recognition) and compared them with thresholds obtained for segmental information (i.e., speech recognition). Implications will be discussed, including the availability of suprasegmental features of speech at levels below the threshold of segmental speech recognition. These results suggest that after speech in a signal becomes unintelligible, communication may still occur through the transmission of suprasegmental information, such as the talker's emotional state.

4pSC20. Experimentally testing the epenthetic bias in loanword adaptation. Zachary Jaggers (Linguist, Univ. of Oregon, University of Oregon Linguist, 161 Straub Hall, Eugene, OR 97403, zjaggers@uoregon.edu)

In loanword adaptation, epenthesis is the favored way to make non-native sound sequences pronounceable, over other options like deletion or substitution (Paradis and LaCharité, 1997). This epenthetic bias is also apparent at the phonetic level, such as the phonologization of excrescent bursts and vocoids as full vowels (Kang, 2003; Davidson, 2007). It is possible that loanword status in and of itself induces this bias, whether or not the source form of interest would be illicit in the speaker's native sound system. Weinberger's (1994) Recoverability Principle suggests that second-language learners prefer to preserve or insert sound material due to less awareness of what may be expendable while retaining word recoverability. The epenthetic bias may therefore hold even for sound sequences available in a speaker's native language but which they consider to be embedded in a word from a foreign language. The current study tests this prediction. Listeners transcribe nonce words manipulated along a [CCVC]-[CəCVC] continuum in which, crucially, both ends of the continuum are licit in their native language. Surrounding speech is manipulated between two framings of the nonce word as either an unfamiliar word in the native language or a word from a foreign language to test whether the latter framing induces a preference for <CVC…> transcription. This shines light on the phonetic roots of a common phonological pattern and how contextually mediated these are.

4pSC21. Prediction of listener perception of place-of-articulation in reduced speech. Megan M. Willi (Commun. Sci. and Disord., California State Univ., Chico, 400 W. 1st St., Chico, CA 95929, mwilli@csuchico.edu), Natasha L. Warner (Linguist, Univ. of Arizona, Tucson, AZ), and Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Previous research on stop consonant production found that less than 60% of the stops sampled from a connected speech corpus contained a clearly defined hold duration followed by a plosive release [Crystal and House, *JASA*(1988)]. How listeners perceive reduced, voiced stop consonant variants is not well understood. The purpose of the current study was to investigate whether an acoustic cue called a relative formant deflection pattern was capable of predicting listeners' perceptions of these approximant-like, voiced stop consonants variants. A new methodology motivated by a computational model of speech production was used to extract relative formant deflection patterns from excised VCV segments from a reduced speech database. Participants listened to a total of 56 excised VCV stimuli containing approximant-like, voice stop consonant variants and performed a force choice test (i.e., /b-d-g/). The agreement between the perceptions predicted by the relative formant deflection patterns and listeners' behavioral performance was compared. The expected relative formant deflection pattern correctly predicted listeners' primary response for percent /b/ and /g/ identifications, but not for listeners' percent /d/ identifications. The implications of these results on a possible invariant acoustic correlate for listeners' perceptions of place-of-articulation information will be discussed.

4pSC22. Speech exposure familiarizes listeners with talkers' vocal identity. Jayden Lee and Tyler K. Perrachione (Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, jaydenl@bu.edu)

Talker identification experiments typically explicitly train listeners to identify voices. Ecologically, however, listeners learn to identify talkers without explicit practice. Here, we investigated whether listeners gain familiarity with voices even when they are not explicitly identifying them. Participants were assigned to three different exposure tasks with feedback, exposing them to identical stimuli but differentially directing their attention between talkers' vocal identity and verbal content: (1) identifying whether the talker and visual cue on each trial matched; (2) discriminating whether the talker was the same as the prior trial; (3) discriminating whether the speech content matched the previous trial. All three groups were then tested on their ability to learn to identify talkers from novel speech content. Critically, we manipulated whether the talkers during this post-test were the same or different than those heard during training. Regardless of exposure

task, listeners were significantly more accurate at learning to identify talkers they had previously been exposed to versus novel talkers. The group that practiced identifying talkers during the exposure phase was only more accurate on exposed talkers. These results suggest that listeners learn talkers' vocal identity during speech perception even if they have not been directed to attend to talker identity.

4pSC23. Investigating the conditions on target-context assimilation in speech sound categorization. Amanda Rysling (Linguist, Univ. of California Santa Cruz, 1156 High St., Dept. of Linguist, Santa Cruz, CA 95064, rysling@ucsc.edu) and John Kingston (Linguist, Univ. of Massachusetts Amherst, Amherst, MA)

Many studies have shown that listeners perceptually differentiate target sounds in categorization tasks from their neighboring context sounds, but some have shown that targets are perceptually assimilated to their contexts. We test the hypothesis that differentiation occurs in context-target order because the context is taken as the criterion for categorizing the target, but assimilation occurs in target-context order because the context's acoustics are parsed as target information. In our experiments, the target was a labial-to-coronal consonant continuum or a front-to-back vowel continuum in VC and CV strings, and the contexts were the other continuum's endpoints. As the second sound in VC or CV, the target differentiated from the preceding context: listeners responded labial or back more often after front vowels and coronal consonants, respectively. With a target V in VC, the target assimilated to the following context: listeners responded back more often before labial consonants. For C in CV, some listeners assimilated the consonant to the following vowel: they responded labial more often before back vowels. Others instead differentiated the consonant from the vowel: they responded coronal more often before back vowels. Follow-up experiments will determine the conditions in which a consonant assimilates to or differentiates from a following vowel.

4pSC24. Speaking rate changes how duration informs phoneme categorization. Andrew Lamont, Rong Yin, Aneesh Naik, and John Kingston (Linguist Dept., Univ. of Massachusetts, 650 N. Pleasant St. ILC 434, Amherst, MA 01003, jkingston000@gmail.com)

Repp *et al. J. Exp. Psychol.* (1978)] reported that for a given duration of fricative noise, a longer silence was required to shift from a fricative to an affricate percept at a slower than a faster speaking rate. We crossed 5 fricative durations (90–208 ms, 29–30 ms steps) by 5 silence durations (0–120 ms, 30 ms steps) by two speaking rates (slow:fast ratio 1.51). Possible responses were grey ship, grey chip, great ship, or great chip. The likelihood of responding ch relative to sh decreased as the fricative lengthened, increased as the silence lengthened, and was more likely at the slow than the fast rate, but neither fricative nor silence duration interacted with speaking rate \— an apparent failure to replicate Repp, *et al.* The likelihood of responding great relative to grey increased with both fricative and silence duration and at the faster than the slower rate. Increasing fricative duration also increased the relative likelihood of responding great more at the slower than the fast rate, but increasing silence duration increased great likelihood less at the slower rate, which indirectly replicates Repp *et al.*, so long as fewer stop responses stand in for fewer affricate responses.

4pSC25. An eye-tracking investigation on the role of categorical perception and acoustic details in the processing of tonal alternations in context. Yu-Fu Chien (Chinese Lang. and Lit., Fudan Univ., Rm. 701, West Wing Guanghuag Bldg., N.220, Handan Rd. Yangpu District, Shanghai, Shanghai 200433, China, chien_yc@fudan.edu.cn) and Jung-Yueh Tu (Ctr. for Int. Chinese Education, Shanghai Jiao Tong Univ., Shanghai, China)

Neutralization is a phenomenon in which two different phonemes are realized as the same sound in certain phonetic environments. In Mandarin, a low-dipping Tone3 is converted to a high-rising Tone2 when followed by another Tone3, known as Third-Tone sandhi. Although previous studies showed statistically differences in F0 between a Sandhi-Tone3 (high-rising) and a Tone2, native Mandarin listeners failed to correctly categorize these two tones in perception tasks (Peng, 2000). The current study utilized the visual-world paradigm in eye-tracking to further investigate whether acoustic details in lexical tone aid lexical access in Mandarin. In the first experiment, we replicated previous studies in that production data of ten disyllabic minimal pairs of Sandhi-Tone3 + Tone3 and Tone2 + Tone3 words showed differences in F0 for the initial tones, but Mandarin listeners' accuracy in identifying them was only around 50%. In the eye-tracking experiment, results showed that proportion of looks to pictures corresponding to Sandhi-Tone3 + Tone3 words was significantly higher when Mandarin listeners heard Sandhi-Tone3 + Tone3 words. A similar pattern was found when auditory stimuli were Tone2 + Tone3 words. The eye-tracking results demonstrated that subtle acoustic details of F0 aid lexical access in a tone language. Mandarin listeners with or without musical training will also be compared.

4pSC26. Auditory integration in the perception of rhoticity. Molly F. Schenker and Anna M. Schmidt (Speech Path. & Aud., Kent State Univ., Speech Pathol. & Audiol., Kent State University, Kent, OH 44242, mdana@kent.edu)

Traditionally, descent of F3 below 2000 Hz at the midpoint has been considered an acoustic correlate for perceived rhoticity. Recent investigations by Heselwood of the auditory integration hypothesis related to rhotics and by Fox and colleagues of "center of gravity" (COG) for stops, diphthongs, and vowels suggested an application to rhotic perception. A resynthesized continuum containing manipulated formant amplitudes to create a high amplitude frequency band descending from above 2000 Hz to below 2000 Hz over 8 steps will be presented to graduate speech pathology students who will judge goodness of rhoticity.

4pSC27. Individual differences in distributional learning for speech: What's ideal for ideal observers? Nicholas R. Monto and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT, nicholas.monto@uconn.edu)

Research demonstrates that efficient speech perception is supported by listeners' ability to dynamically modify the mapping to speech sounds to reflect cumulative experience with talkers' input distributions. Here we test the hypothesis that higher-level receptive language ability is linked to adaptation to low-level distributional cues in speech input. Listeners completed two blocks of phonetic categorization for stimuli that differed in voice-onset-time (VOT), a probabilistic cue to the voicing contrast in English stop consonants. In each block, two distributions were presented, one specifying /g/ and one specifying /k/. Across the two blocks, variance of the input distributions was manipulated to be either narrow or wide, reflecting distributions that were relatively more to relatively less consistent, respectively, in terms of how VOT cued the voicing contrast. As predicted by ideal observer computational frameworks, the participants in the aggregate showed steeper identification slopes for consistent compared to inconsistent input distributions. However, the magnitude of learning showed wide individual variability, which was predicted by receptive language ability as measured using standardized assessments. Individuals with poorer receptive language scores showed diminished distributional learning due to a failure to capitalize on consistent input distributions; instead, their perceptual decisions showed instability even the face of acoustic-phonetic certainty.

4pSC28. A deep neural network approach to investigate tone space in languages. Bing'er Jiang, Tim O'Donnell, and Meghan Clayards (McGill Univ., 1085 Dr. Penfield, Montreal, QC H3A 1A7, Canada, binger.jiang@mail.mcgill.ca)

Phonological contrasts are usually signaled by multiple cues, and tonal languages typically involve multiple dimensions to distinguish between tones (e.g., duration, pitch contour, and voice quality, etc.). While the topic has been extensively studied, research has mostly used small datasets. This study employs a deep neural network (DNN) based speech recognizer trained on the AISHELL-1 (Bu *et al.*, 2017) speech corpus (178 hours of read speech) to explore the tone space in Mandarin Chinese. A recent study shows that DNN models learn linguistically-interpretable information to distinguish between vowels (Weber *et al.*, 2016). Specifically, from a low-dimensional Bottleneck layer, the model learns features comparable to F1 and F2. In the current study, we propose a more complicated Long Short-

Term Memory (LSTM) model—with a Bottleneck layer implemented in the hidden layers—to account for variable duration, an important cue for tone discrimination. By interpreting the features learned in the Bottleneck layer, we explore what acoustic dimensions are involved in distinguishing tones. The large amount of data from the speech corpus also renders the results more convincing and provides additional insights not possible from studies with more limited data sets.

4pSC29. Bidirectional decay of auditory memory traces for pitch in speech sounds. Zhanao Fu (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, 1265 Military Trail, Scarborough, ON M1C 1A4, Canada, zhanao.fu@mail.utoronto.ca) and Philip J. Monahan (Dept. of Linguist, Univ. of Toronto, Toronto, ON, Canada)

Previous studies have shown human listeners have greater detection sensitivity to pitch increment than decrement in successive sounds. Assuming deviance detection is based on the comparison between the memory trace of a recent stimulus and the neural representation of a new stimulus, one hypothesis is that this differential sensitivity between increment and decrement is caused by the downward decay of pitch's memory trace. Under the same assumption, the present study found bidirectional—as opposed to the predicted unidirectional—decay of memory traces for pitch in speech sounds by measuring listeners' sensitivity to pitch change over varying time intervals with an AX discrimination task. Three properties in the AX task were randomly sampled from preset ranges: (1) f_0 of the A token (163:320 Hz), (2) difference between the f_0s of A and X (-30:30Hz), and (3) the inter-stimulus interval (ISI; 0:3 s). We found when the stimuli were in the lower portion of the speaker's pitch range, listeners were less sensitive to pitch increments at larger ISIs. Meanwhile, when the stimuli were in the higher pitch range, listeners were less sensitive to pitch decrements at larger ISIs. These results suggest memory traces for pitch in speech sounds decay toward a center pitch.

4pSC30. Directionality in sound change from asymmetries in acoustic distribution. Ollie Sayeed (Dept. of Linguist, Univ. of Pennsylvania, 3401 Walnut St., Philadelphia, PA 19104, sayeedo@sas.upenn.edu)

Following the work of John Ohala, historical sound changes are thought to take place by misperception of the input on the part of the listener. Any account of sound change based on misperception, though, faces a paradox: if X sounds like Y, Y should also sound like X, and yet we often see sound changes that are only attested in one direction. A potential solution is to think of phonetic categories as distributions in acoustic space, and so asymmetries in sound change ($X > Y$, $*Y > X$) come from asymmetries in the spread of the distribution of X and Y. If X is a very variable phonetic category with a thick-tailed distribution, a high proportion of its tokens should cross the perceptual boundary and be misperceived as Y; if Y has a narrow distribution, only a very small proportion of its tokens should be perceived as X. We predict that unidirectional sound changes should involve a change from a high-variance to a low-variance category. This experiment tests a case study of asymmetric nasal place assimilation (VnpV > VmpV, *VmpV > VnpV) on a sample of six speakers in three vowel contexts. In the contexts a_a and i_i, the sequence np before the change had a higher-variance distribution of F2 transition (a cue to nasal place) than the sequence mp after the change. In u_u, the sequence after the change had the higher variance. These results give partial support to our hypothesis.

4pSC31. On the articulation between acoustic and semantic uncertainty in speech perception: Investigating the interaction between sources of information in perceptual classification.. Olivier Crouzet (Laboratoire de Linguistique de Nantes (LLING), Université de Nantes / CNRS, Chemin de la Censive du Tertre, Laboratoire de Linguistique de Nantes (LLING), Nantes 44312 Cedex, France, olivier.crouzet@univ-nantes.fr) and Etienne Gaudrain (Ctr. de Recherche en NeuroSci. de Lyon – CRNL, CNRS / INSERM / Université Lyon 1, Groningen, The Netherlands)

Listeners processing speech signals have to deal with two main classes of uncertainty occurring in the vicinity of a given speech segment: both acoustic properties of the contextual environment (Ladefoged and Broadbent, 1957; Sjerps and McQueen, 2013) and lexical hypotheses based on word co-occurence probabilities or semantic relations (e.g., Connine, 1987; Gow and Olson, 2015) may affect the interpretation of a given sound. We investigate this issue by independently manipulating (1) semantic relationships between words using word embeddings estimations and (2) acoustic relations between a contextual part and the final word in the sentence. Based on word pairs that contrast on their vowel target only (e.g., french "balle" versus "belle", pronounced /bal/ vs. /bEl/ – eng. "ball" versus "beauty"), 3 types of sentences are generated: (1) a sentence that would semantically "prime" the word /bal/ ("Le joueur a dévié la", eng. "The player deflected the"), (2) a sentence that would favour the word /bEl/ "Le prince a charmé la", eng. "The prince charmed the"), and (3) a semantically incongruous sentence in both cases "Le journaliste a découvert la", eng. "The journalist discovered the." Listeners are presented with fully ambiguous final words (acoustically located between, e.g., /bal/ and /bEl/) in contexts where semantic influence varies (sentence-types 1/2/3) and is balanced with acoustic manipulations of formant frequencies favouring one word or the other. This will provide cues to modelling how both sources of entropy alter speech perception.

4pSC32. Real-time auditory feedback perturbation of German quantity contrasts. Miriam Oschkinat (Inst. of Phonet. and Speech Processing, Ludwig Maximilan Univ. of Munich, Schellingstraße 3, 80433, Munich 80433, Germany, miriam.oschkinat@phonetik.uni-muenchen.de), Eva Reinisch, and Philip Hoole (Inst. of Phonet. and Speech Processing, Ludwig Maximilan Univ. of Munich, Munich, Bavaria, Germany)

Online auditory feedback (OAF) perturbations have reviewed much about the interplay between acoustic and sensorimotor information during speech production. For spectral manipulations (e.g., formant frequencies), it was shown that people are sensitive to OAF, mainly reacting with a compensation in the opposite direction to the perturbation. This study investigates German speakers' reaction not to spectral but temporal OAF manipulations for the vowels /a/ and /a:/, a phoneme contrast that is realized as a quantity contrast without strong additional spectral cues. Participants were asked to produce the German words *Stab* (/ʃtaːp/ "pole") and *Staat* (/ʃtaːt/ "state") where the vowel was compressed in real-time, or *Stamm* (/ʃtam/ "trunk") and *Stadt* (/ʃtat/ "city") where the vowel was lengthened. While *Staat* and *Stadt* form a minimal pair in German, *Stamm* and *Stab* do not have lexical neighbours. Results showed compensatory responses in the opposite direction to the manipulation for *Staat*, *Stab* and *Stamm* with larger effects for *Staat* (with the lexical neighbour) than *Stab* (without lexical neighbour). Thus, participants react to manipulations of temporal feedback in a similar manner to spectral perturbations. These findings give more precise insight into the link between perception and production in the online-processing of the temporal structure of speech.

4pSC33. Adaptive measurement of crossover frequencies for intelligibility prediction. Nathaniel A. Whitmal (Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003, nwhitmal@umass.edu)

In SII theory, frequencies where speech spectra can be divided into two equally-intelligible subbands are called crossover frequencies. These frequencies play a crucial role in SII calculations, and also designate spectral regions that contain important speech recognition cues. Typically, crossover frequencies are found by measuring psychometric curves for speech processed by a series of low-pass and high-pass filters, and then finding the two curves' intersection: an inefficient, time-consuming process. The present study introduces an up/down quantile estimation algorithm that adaptively steers filter cutoff frequencies toward the crossover frequency. Changes in cutoff frequency are governed by comparisons of block trials for low-pass and high-pass filtered speech that meet theoretical requirements for convergence toward the crossover frequency. Preliminary results for trials with nonsense syllables show that the proposed method's estimates match those obtained in published trials using the conventional method. Applications in SII measurements and speech recognition cue measurement will be discussed.

4pSC34. Lexically dependent estimation of acoustic information in speech II: Minimal pair confusability. Charles Redmon and Allard Jongman (Dept. of Linguist, Univ. of Kansas, 1541 Lilac Ln., Rm. 427, Lawrence, KS 66046, redmon@ku.edu)

We aim to develop a framework for the analysis of phonetic contrast systems that is fundamentally lexical and does not depend on assumptions of inventory homogeneity and independence of distribution in words and higher-order systems. Previously (Redmon and Jongman, 2018, *JASA*) we reported results of an open-class identification experiment on a 240-word sample of the 26,793-word single-speaker database in Tucker *et al.* (2018). Here, we present results of the second experiment in the project: a 2AFC task where the choice set is limited to obstruent-contrastive minimal pairs. This task forms the opposite end of a continuum from least restricted utilization of acoustic or higher-order information (Exp. 1), to localized attention to a particular contrast in the lexicon. Just as the first experiment provided estimates of a lower bound on listeners' sensitivity to different cues in the signal, the results of this experiment provide an upper bound on those estimates. Participants were presented with 480 stimuli balanced between contrastive obstruents in #CV, VCV, and VC# positions. The results were then used to determine network edge weights on a phonological lexicon on the model of Vitevitch (2008), which emphasizes the interaction between acoustic features, neighborhood topology, and higher-order information in the lexicon.

4pSC35. What *can* make clear speech clear: Lessons learned from the Ferguson Clear Speech Database. Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu)

Extensive acoustic and perceptual analyses have been carried out on the materials from the Ferguson Clear Speech Database (FCSD), which was recorded at Indiana University in 2002. The FCSD consists of 41 untrained talkers reading 188 sentences under instructions first to speak in a manner "as much like your normal conversational style as possible" and later to "speak clearly, so that a hearing-impaired person would be able to understand you." My intent in developing the FCSD was to exploit the expected wide acoustic and perceptual variability among the talkers and use a talker-differences approach to answer the question, "What makes clear speech clear?" In this presentation, I will summarize data from studies of vowel intelligibility, word intelligibility, and perceived sentence clarity along with global and fine-grained acoustic analyses, and discuss how all of these measures are related across the 41 talkers. My hope is that this birds-eye view of the FCSD data will reveal subgroups of talkers in which the talkers adopted certain "profiles" of clear speech acoustic changes that yielded specific helpful perceptual changes. If time permits, I will also review data on perceived talker indexical properties and how they change when talkers speak clearly.

4pSC36. A replication of a test of the metrical segmentation strategy in spoken word recognition. Natasha L. Warner, Seongjin Park (Univ. of Arizona, Dept. of Linguist, University of Arizona, Tucson, AZ 85721, seongjinpark@email.arizona.edu), James M. McQueen (Donders Inst., Radboud Univ., Nijmegen, The Netherlands), Richard A. Southee, Dongdong Zhang, and Iris Lin (Univ. of Arizona, Tucson, AZ)

Norris *et al.* (1995) tested the Metrical Segmentation Strategy (MSS; Cutler and Norris, 1988) as part of the spoken-word recognition model Shortlist. We replicate their study in a different dialect of English, with a different population and items. Norris *et al.* used a word-spotting task, in which listeners had to spot words within speech (e.g., stamp in [stæmpɪdʒ]). Target words were CVCC like champ or CVC like done, and were followed by a full vowel (champ in/tʃæmpoʊʃ/, done in /dʌnʤeɪb/) or a reduced vowel (champ in /tʃæmpəʃ/, done in /dʌnʤəb/). The original study found different behavior for CVCC versus CVC targets, with the results suggesting that listeners hypothesize a word onset at the start of a full-vowel strong syllable (the MSS). Doing so makes it harder to detect champ when it is followed by a full vowel than a weak vowel because the full vowel leads the listener to think the /p/ is the onset of the following word, while the following vowel has little influence for done, where the equivalent consonant is not part of the word. The results for the current study (underway) will show whether these effects generalize across English dialects, listener populations, and words.

4pSC37. Facilitation of speech processing by both expected and unexpected talker continuity. Yaminah D. Carter, Alexandra M. Kapadia, Sung-Joo Lim, and Tyler K. Perrachione (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ycarter@bu.edu)

Speech processing is faster for one continuous talker than mixed talkers. However, it is unknown whether listeners' expectations about talker continuity affect this facilitation. We measured response times during three speeded word identification experiments that manipulated listeners' expectations about talker continuity. First, we manipulated expectations about talker continuity by presenting words in pairs where both words were frequently produced by the same talker (talker-repeat trials) and rarely by different talkers (talker-change trials), or vice-versa. Word identification was faster in talker-repeat trials than talker-change trials, with equal facilitation from both expected and unexpected talker continuity. Unexpected talker changes did not slow processing more than expected changes. Second, a control experiment demonstrated that listeners' expectations about repetitions of the word itself did affect word identification speed. Third, listeners identified words in conditions with one talker, two talkers presented randomly, or two alternating talkers. Word identification was faster whenever the talker was repeated compared to when the talker switched between trials, even if listeners could perfectly predict the talker switch (i.e., alternating-talker condition); talker continuity also facilitated word identification in the random condition. These results provide converging evidence that talker continuity facilitates speech processing in an automatic, feed-forward way, irrespective of listeners' expectations.

4pSC38. Effects of type, token, and talker variability in speech processing efficiency. Alexandra M. Kapadia, Jessica Tin, and Tyler K. Perrachione (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, akapadia@bu.edu)

Phonetic variability across talkers imposes additional processing costs during speech perception, evident in performance decrements for mixed- versus single-talker speech. However, within-talker phonetic variation across different utterances is another, relatively unexplored source of variability in speech, and it is unknown how processing costs from within-talker variation compare to those from between-talker variation. Cognitive consequences of talker variability are also mostly measured from two-alternative forced-choice tasks, whereas naturalistic speech processing occurs in a much larger decision space. Do talker-variability effects scale when both the stimuli and the decision space are more complicated? Here, we measured response times in a speeded word identification task that factorially manipulated three dimensions of speech variability: number of talkers (one versus four), number of target word choices (two versus six), and number of talker-specific exemplars per word (one versus eight). Across all eight experimental levels, larger decision spaces led to significantly slower word identification. Word identification was also slower in conditions with mixed talkers and conditions with multiple exemplars. This pattern of interactions suggests complex processing relationships between type, token, and talker variability and provides preliminary evidence for how both within- and between-talker variability impose additional processing costs in speech perception.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will meet starting at 4:30 p.m.

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

Committees meeting on Tuesday are as follows:

Engineering Acoustics (4:30 p.m.)	McCreary
Signal Processing in Acoustics (4:30 p.m.)	Beckham
Acoustical Oceanography	McCreary
Animal Bioacoustics	Clements
Architectural Acoustics	French
Musical Acoustics	Breathitt
Physical Acoustics	Jones
Psychological and Physiological Acoustics	Carroll Ford
Structural Acoustics and Vibration	Stopher

Committees meeting on Wednesday

Biomedical Acoustics	Nunn

Committees meeting on Thursday

Computational Acoustics (4:30 p.m.)	Clements
Noise	Segell
Speech Communication	Carroll Ford
Underwater Acoustics	McCreary

Session 5aAA

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

Klaus Genuit, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

Kenneth W. Good, Cochair
Armstrong, 2500 Columbia Ave., Lancaster, PA 17601

Chair's Introduction—8:00

Invited Papers

8:05

5aAA1. Restaurant acoustics—A challenge for consultants. Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

A good restaurant should be a place where you can enjoy food, drinks, and talking with friends. This means besides the quality of the dishes, wines, etc., the acoustical comfort in the restaurant is an important parameter. But how to describe a good acoustical quality of a restaurant, what are the parameters, how to measure, and how to evaluate the situation? Not only is the level an indicator but others like privacy and speech intelligibility are also indicators. As the overall perception and evaluation of the event inside a restaurant is strongly dependent on context besides measurement and analysis, interviews of the involved people are requested like it is recommended within the new ISO TS 12913-2 standard for soundscape. An overview of different factors influencing the restaurant acoustics and methods exploring the sound will be given.

8:25

5aAA2. What is "typical" restaurant design today? Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

More and more patrons and restaurant owners are acknowledging that acoustic comfort contributes to an enjoyable experience. So, what is the "typical" design and the "typical" acoustics of restaurants and similar hospitality spaces today? This paper will explore basic information for over 50 restaurants and hospitality spaces. Included in the data, we will consider room size, ceiling height, surface, and basic room acoustic. This will present a cross-section of the current industry designs and issues that the acoustic community faces today.

8:45

5aAA3. Restaurant acoustics—Unconducted and sometimes unintended orchestrations—An overview. Wade Bray (HEAD Acoust., Inc., 6964 Kensington Rd., Brighton, MI 48116, wbray@headacoustics.com)

The modern restaurant, such as the modern concert hall, is a societal orchestration on many levels including and involving acoustics. The restaurant overlays a business approach, diner tastes, architecture, and other socio-economic factors in superimposed circles of interaction thought to be planned but in action often unanticipated with unwitting consequences. The n-dimensional influences of restaurant acoustics have distinct parallels with the n-dimensional influences of concert hall acoustics. In a concert hall, it is not enough to have architectural beauty, a quiet background and good acoustics in the audience area if onstage acoustics do not support the musicians as individuals in an ensemble. If players must expend concentration on coping instead of elevating unhindered into their art, the quality the audience desires of the music cannot be reached. But if both onstage and audience area acoustics are good, the music soars, audiences keep coming and paying, all participants reap positive rewards, and the organization is successful and self-perpetuating. In this overview, we suggest that the influence of a restaurant's acoustics on its success/satisfaction outcome is perhaps even more complex with more human dimensions and less known than that of the concert hall but that observing and considering these parallels may be beneficial.

9:05

5aAA4. Restaurants part 3: The acoustics of dining spaces. Keely Siebein, Gary W. Siebein, Hyun Paek, Gary Siebein, Marylin Roa, Jennifer R. Miller, and Matthew Vetterick (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, ksiebein@siebeinacoustic.com)

Restaurants are spaces where people gather to eat, talk, share, and spend time. They are also facilities where food is prepared and served to others. Restaurants have garnered much attention over the last several years as having high sound levels with some customers having difficulty in communication among patrons and staff. However, spaces such as cafeterias, dining rooms in country clubs, retirement centers, and hospitals serve similar functions as restaurants and suffer from similar acoustic issues. Acoustic measurements of alpha bar, reverberation time, reflected sound energy, noise reduction, and STI were made in various types of dining spaces that have required acoustic treatment. Data from these types of spaces have been added to the data previously collected at restaurants, to provide general information about the acoustics of dining spaces. The results show that dining spaces can be broken into similar treatment categories, regardless of the specific type of space. The sample of rooms was also analyzed by architectural and interior design style to identify design trends that affect the acoustical metrics in dining spaces.

9:25

5aAA5. An innovative approach in data collection for restaurant soundscape study. W. M. To (Macao Polytechnic Inst., Macao, Macao) and Andy Chung (Macau Instituto de Acústica, Macao, Macao, Macao, ac@smartcitymaker.com)

Crowdsourcing is one of the common approaches in collecting sonic and human perception information for studying restaurant soundscape. However, it has not been easy to find a good reason to motivate people so as to obtain their support continuously in data collection. An innovative approach has been developed and tried in various locations in Asia. This paper presents the initial findings.

9:45–10:00 Break

Contributed Papers

10:00

5aAA6. A restaurant noise study of continuous and transient noise phenomena. Tiffany A. Bixler (Phys. Dept., U.S. Naval Acad., 109 Archwood Ave., Glen Burnie, MD 21061, tiffanybixler7@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

Two restaurants that the author (Bixler) is familiar with are compared in this noise study. Restaurant 1 is a moderately busy (with varying crowds) sports bar that is considered "Americana." Restaurant 2 is a very busy (always crowded) Texas themed sports bar. Measurements use a RadioShack model 3300099 Digital Sound Level Meter and a SoundMeter X (Version 10.2.2) (application by faberacoustical.com that can be downloaded free from the App Store onto an i-Phone). A professional analysis "Pro tool set" is available and includes a level meter, dosimeter, octave analyzer, and a recorder. These restaurants have interesting areas for studying both continuous and transient noise phenomena. Noise measurement locations include: the dishwasher's sink (sounds of dishes being washed), the surrounding grill area (sounds of cooking food and moving pans), the bar, the main dining area, and the high-top tables near the bar (to measure the rise and fall of sound levels as conversations ensue). Noise measurements include the prep area, the back sink, behind the bar, host stand, foyer, and main seating areas. Transient noise and sound levels averaged (between 15 s and 1 min) will be presented for different times for slow periods, mediums crowds, and full capacity seating.

10:15

5aAA7. Distributions of reverberation time measurements from restaurants of different Layouts. Victoria R. Anderson (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 4621 A St., Omaha, NE, 68182, Lincoln, NE 68510, vranderson@unomaha.edu), Kieren H. Smith, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE, 68182, Lincoln, NE)

The purpose of this research is to determine how reverberation time differs spatially in restaurants with different seating arrangements such as spaces with one main dining room versus spaces with several connected dining areas branching off a main area. The physical layouts of a number of restaurants were spatially recorded, and impulse responses were taken at set interval locations throughout the unoccupied restaurants. Reverberation times have been calculated from the impulse responses and spatially mapped throughout the rooms in order to visually understand the distributions. The average reverberation time and standard deviations in each restaurant are presented and compared to similar data calculated previously for classrooms (Peng *et al*., 2012). Sound levels have also been logged from each restaurant during occupied times, and relationships between sound levels in an area and local reverberation time are explored.

Invited Paper

10:30

5aAA8. Characterization of restaurant soundscapes based on logged sound levels and occupancy measurements. Kieren H. Smith, Victoria R. Anderson, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Peter Kiewit Inst., Univ. of Nebraska-Lincoln, Omaha, NE 68182, kierenhs@gmail.com)

Sound levels in a restaurant depend on a variety of factors including the restaurant's room shape, materials, seating layout, and the levels of ambient and human-produced noise. To better understand the characteristics of restaurant acoustics in real world settings, an *in situ* study has been undertaken in a sampling of sit-down dining venues. Assorted geometric room characteristics, impulse responses, and background noise data were gathered from the spaces. Sound levels were also logged in the restaurants during operating hours simultaneously with occupant count and density using thermal imaging. The gathered data are analyzed to characterize the soundscapes of some existing restaurants. In particular, the research aims to understand how sound levels increase with occupancy and how specific architectural and design features such as seating style and density may contribute to experienced sound levels.

10:50–11:10 Panel Discussion

Session 5aAB

Animal Bioacoustics, Signal Processing in Acoustics, Acoustical Oceanography, Speech Communication, and Underwater Acoustics: Understanding Animal Song

John Hildebrand, Chair
Scripps Institution of Oceanography, University of California San Diego, Mail Code 0205, La Jolla, CA 92093

Invited Papers

8:00

5aAB1. The characteristics of song: An overview for marine mammals. John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, UCSD-0205, La Jolla, CA 92093, jhildebrand@ucsd.edu)

Animal song is often considered to have the following characteristics: (1) produced by males, (2) territorial—related to ownership of an area, (3) stereotyped, (4) seasonally produced, and (5) involved in reproductive displays. The extent to which marine mammals fit these criteria is considered for a variety of species. The occurrence of female song and the lack of clear territoriality are examples that challenge the above model for sexual-selection of singers. Likewise, the factors driving long-term trends in song are not well understood. New perspectives are needed to understand the mechanisms driving marine mammal song.

8:25

5aAB2. Evidence of synchronous chorusing in North Atlantic minke whales (*Balaenoptera acutorostrata*. Michael A. Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

Many arthropods are known to chorus acoustically. Some families such as the *Gryllidae* are known to synchronize their chirping, and other families such as the *Cicadidae* are known to buzz together or pulse in groups. In the *Gryllidae,* synchronization is facilitated by way of corollary discharge that "blanks" the auditory nerve concurrent to stridulation syllables. This permits the individual animal to "not hear" itself while chirping and to hear nearby conspecifics if they are not in sync. Thus, when animals are synchronized, none of them hear anything, unless one falls or phases out of sync. There are many conjectures as to why they synchronize. It is clear that in doing so, they establish an acoustic community. The larger sound-field also ambiguates the location of individual animals to potential predators. There is some evidence that conspecific communication occurs within the chorus which may convey local conditions (such as the presence of a predator) to the acoustic community. Many rorquals (blue, fin, sei, Brydes, and minke whales) pulse in the low and infrasonic frequency ranges. This paper will present and evaluate recorded evidence which might suggest that minke whales may synchronize their pulsing with others—potentially forming an acoustic community with conspecifics.

8:50

5aAB3. Song structure and sex-specific features in the indris. Marco Gamba, Daria Valente, Chiara De Gregorio, Anna Zanoli (Life Sci. and Systems Biology, Univ. of Torino, Via Accademia Albertina 13, Torino 10123, Italy, marco.gamba@unito.it), Rose Marie Randrianarison (Life Sci. and Systems Biology, Univ. Of Torino, Antananarivo, Madagascar), Valeria Torti, and Cristina Giacoma (Life Sci. and Systems Biology, Univ. of Torino, Torino, Italy)

In the dense tropical rainforests of Madagascar, visual communication is impeded by obstacles. Lemurs use scent marking to communicate at a short distance and loud calls to communicate at a long range. The Indri (Indri indri) is a diurnal primate that emits choruses of three distinct types. The song types are essential in advertising position within the group territory, in deciding the sorts of aggressive group encounters, and in maintaining cohesion between animals dispersed during feeding. A detailed examination of the advertisement songs showed that three main parts constitute them. The last, most consistent of these parts, showed phrases consisting of units emitted with a descending frequency. The rhythm of these units changed significantly between phrases of different durations and the sexes. We also found that the unit structure may provide conspecifics with information on the individual identity of the emitter and that male songs, but not female's, are more similar to those of their parent of the same sex than to other indris. The song of the indris is an effective way to transfer information at a long distance and may transmit sex and individual identity information.

9:15

5aAB4. Mosquito hearing is the most sensitive among arthropods—But is the sound level of a male swarm loud enough to be picked up by the female's particle-velocity sensor? Lionel Feugère (MIVEGEC, IRD, CNRS, Univ. Montpellier, Montpellier, France and Natural Resources Inst., Univ. of Greenwich, Central Ave., Chatham Maritime ME4 4TB, United Kingdom, lionel.feugere@ird.fr), Gabriella Gibson (Natural Resources Inst., Univ. of Greenwich, Chatham, Kent, United Kingdom), and Olivier Roux (MIVEGEC, IRD, CNRS, Univ. Montpellier, Montpellier, France)

Males of many mosquito species aggregate in station-keeping swarms, waiting for the arrival of conspecific females to mate with. We test whether audition could be used by a female to locate male swarms and to assess whether the males are conspecific. The sound level resulting from thousands of wing flaps could be loud enough to be heard at a long range (~1 m) via the antennal flagellum (particle velocity sensor, primarily designed for close-range communication). A mosquito hears a conspecific by adjusting its own wing-beat frequency so that the difference tone between its own and the opposite-sex frequencies falls into a narrow band to which the auditory organ is tuned. Indeed, the antennal flagella produce distortion products resulting in difference tones of the nearby soundscape. Swarms of males were recorded and played-back to females in a 2-m-sided flight chamber. The natural sounds of the males of two species (*Anopheles coluzzii* and *A. gambiae*) and related synthetic sounds were played at different sound levels to individual free-flying *A. coluzzii* females. The mosquitoes' responses were investigated by analysing changes in three-dimensional-tracked flight trajectories and wing-beat frequencies. The results show that (1) females do respond to the sound of swarming males, (2) a qualitative difference between female and male behaviour, (3) a quantitative effect of the sound stimulus of conspecific males, and (4) verification of previous results suggesting the importance of the first harmonic of their wing beats in mosquito acoustic communication.

9:30

5aAB5. Deep learning for underwater noise classification. Johnny L. Chen, Sarah Nguyen, Jason M. Trader, Andrew Moore (Appl. Res. in Acoust., LLC, 209 N. Commerce St., Ste 300, Culpeper, VA 22701-2780, johnny.chen@ariacoustics.com), and Jason E. Summers (Appl. Res. in Acoust., LLC, Washington, District of Columbia)

Deep-learning models have surpassed many computer-vision benchmarks and groups such as Google have begun to investigate similar methods for understanding underwater data. Understanding underwater soundscapes is critical to many applications such as assessing the impacts of anthropogenic noise on sea life and monitoring the health and biodiversity of the ocean. In this work, we present a computer-vision approach for classifying audio signals to distinguish whale sounds—especially, vocalizations from mysticetes—from other sources of sound in the underwater soundscape, such as ships. This is a challenging problem due to wide variation in ambient background noise, sensor configuration and properties, and whale vocalization patterns within and across species. Here, we adapt deep convolutional neural networks (CNN) to analyze spectral patterns of common noise sources and demonstrate robust performance on a dataset of ambient noise derived from multiple open-source databases including whale vocalizations from eight species and shipping noise from over ten platforms observed across multiple environments with a variety of sensors. Performance of the network is characterized in terms of classification accuracy and generalizability as a function of CNN hyperparameters and training architecture. With a CNN trained from scratch, we analyze the learned features of the classification decisions of the network by adapting several visualization techniques from the computer-vision domain.

FRIDAY MORNING, 17 MAY 2019 NUNN, 8:00 A.M. TO 11:30 A.M.

Session 5aBA

Biomedical Acoustics: General Topics in Biomedical Acoustics III

Alfred C. Yu, Cochair
University of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada

Julianna C. Simon, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201E Applied Sciences Building, University Park, PA 16802

Contributed Papers

8:00

5aBA1. Electroacoustic tomography system using ultra-short electric filed excitation source induced acoustic signals. Ali Zarafshani (Eng. and Design, Easter Washington Univ., 319 G Computing and Eng. Bldg., Cheney, WA 99004, azarafshani@ewu.edu), Bin Zheng, and Liangzhong Xiang (Medical Imaging, School of Elec. and Comput. Eng., Univ. of Oklahoma, Norman, OK)

Application of an ultra-short, high-intensive electric field has been gaining momentum to many advanced clinical techniques for treating deep-seated tumors based on the electroporation technique. Real-time monitoring is essential for such high impact clinical techniques. Real-time monitoring of electroporation-based medical interventions could be based on the electroacoustic tomography, where the electric field applied for the electroporation process leads to induce acoustic signals based on the flow of electrical current inside the target tissue and then the effects of rising temperature on acoustic pressure. A μs-to-ns high-voltage electric-pulse excitation source is connected to two electrodes separated by a distance of 200μm, where mounted into the conductive media to simulate tissue. An unfocused ultrasound transducer with a central frequency of 500 kHz is used to acquire real-time acoustic signals induced by the electric field. Various conductive media were studied using this pulsed excitation source to induce corresponding acoustic signals, forming two-dimensional tomography images. Results indicate the feasibility of the enhancing the electroacoustic

tomography system that used the high-voltage, ultra-short pulsed excitation source for clinical processes as an imaging guidance to real-time, *in situ* monitoring for the electroporation-based techniques.

8:15

5aBA2. Evaluation of a passive super-resolution beamforming technique for B-mode imaging. Anil Agarwal and Michael Oelze (Univ. of Illinois at Urbana-Champaign, 405 N Mathews Ave., Urbana, IL 61801, aragarw2@illinois.edu)

Improving spatial resolution of ultrasonic scanners would allow improved imaging performance. A lateral resolution is limited by diffraction and the width of the beam used to interrogate the medium. Techniques for improving the lateral resolution beyond the diffraction limit have utilized dilute concentrations of microbubbles (MBs) to localize MB locations. These approaches enable the mapping of microvasculature but do not allow improvements in general B-mode imaging. Null Subtraction Imaging (NSI) is a passive nonlinear image processing scheme that constructs images from the nulls in the beam pattern of an array rather than the main lobe. In NSI, envelopes reconstructed with different apodizations are combined to form an image with an improved apparent lateral resolution. NSI is computationally efficient compared to adaptive beamforming approaches and does not need MBs. Experiments have demonstrated improvements in a resolution of over 100 times compared to conventional delay-and-sum beamforming with no apodization. Engineering trade-offs associated with combining different apodization functions, angular steering, and interpolation were assessed using the NSI technique. The utilization of the NSI for specific imaging tasks was also evaluated. The results of the study suggest that NSI could provide improved performance for the detection of small microcalcifications, which has application in detecting and identifying malignant cancers.

8:30

5aBA3. Tissue Doppler imaging to detect muscle fatigue. Joseph Majdi and Siddartha Sikdar (George Mason Univ., 4400 University Dr., Fairfax, VA 22030, jmajdi2@masonlive.gmu.edu)

Functional electrical stimulation (FES) is often used for rehabilitation in movement disorders and in assistive devices such as exoskeletons. However, FES can rapidly cause muscle fatigue, which limits the induced force production. At present, there exists no reliable, real time indicator for FES-induced muscle fatigue. We believe that functional muscle physiology associated with muscle fatigue can be inferred from ultrasound imaging. In this study, we utilized tissue Doppler imaging (TDI) to quantify FES-induced twitch responses in the gastrocnemius muscle, at baseline and after inducing fatigue through repeated voluntary isometric contractions. We estimated muscle velocities using M-mode TDI to quantify differences in the twitch response before and after fatigue. Preliminary results indicate that fatigue induces a higher muscle acceleration during twitch, and the muscle contracts for a longer duration. These results could potentially be used as a real-time indicator for muscle fatigue. We are investigating the use of such a system integrated into an external hybrid walking exoskeleton that can switch from FES-induced force generation to external motors for force generation once the muscle fatigues. Furthermore, it may be possible to replace TDI imaging with a wearable single-element continuous wave Doppler instrument for these measurements, reducing computational complexity and power requirements.

8:45

5aBA4. Interpretability and generalizability of a one-dimensional convolutional neural network method for hepatic steatosis characterization. Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 North Wright St., Urbana, IL 61801, han51@uiuc.edu), Yingzhen N. Zhang, Michael P. Andre (Liver Imaging Group, Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), Rohit Loomba (NAFLD Res. Ctr., Div. of Gastroenterology, Dept. of Medicine, Univ. of California at San Diego, La Jolla, CA), John W. Erdman (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois at Urbana-Champaign, Urbana, IL), Claude B. Sirlin (Liver Imaging Group, Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), and W. D. O'Brien Jr. (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Nonalcoholic fatty liver disease (NAFLD) affects 25% of the population globally. We developed a one-dimensional convolutional neural network (1-D CNN) method for the ultrasound tissue characterization using radio-frequency (RF) data and demonstrated its potential for the liver fat classification and quantification. We investigate herein the interpretability and generalizability of the method to understand why it works, and whether the performance is affected by settings, transducers, and platforms. We studied under various conditions the performances of 1-D CNN for predicting steatosis using magnetic resonance imaging-estimated proton density fat fraction (MRI-PDFF) as reference (steatosis: PDFF $>$ 5%). Three datasets were used, each containing ultrasound RF data acquired from adults and same-day MRI-PDFF estimates: (1) 200 normal and NAFLD participants scanned using the Siemens S2000 ultrasound system with the 4C1 transducer (1–4 MHz); (2) 87 participants with known/suspected NAFLD scanned using Siemens S3000 with the 4C1 and/or 6C1HD transducers (1.5–6 MHz); and (3) 46 participants with known/suspected NAFLD scanned using GE Logiq e9 with the C1-6 transducer (1–6 MHz). The 1-D CNN method is generalizable among various instrumentation settings (e.g., focal depth, time gain compensation), betweentransducers of similar frequencies, between platforms, but not between the fundamental and tissue harmonic image modes. [Work supported by R01DK106419.]

9:00

5aBA5. A study on the improvement of a brain concentration using acoustic characteristics of sound. Seonggeon Bae (Div. of Comput. Media Information Eng., Kangnam Univ., 40, Kangnam-ro, Giheong-gu, Youngin-si 446-702, Gyeonggi-do, South Korea, sgbae@kangnam.ac.kr) and Myungjin bae (TeleCommun. and Information, Soongsil Univ., Seoul, Korea)

It has been studied variously using acoustic features. The purpose of this study is to investigate whether the characteristics of acoustic signals improve the brain concentration. Especially, the phenomenon that occurs in a tedious and repetitive environment is analyzed by using acoustic characteristics of sound. This characteristic was measured by 20 persons in general, and the concentration of brain was compared and analyzed.

9:15

5aBA6. On a recovery of hoarseness by reflecting the amplified voice. Myungsook Kim and Myungjin Bae (Commun. Eng., Soongsil Univ., Sangdo-ro 369, Seoul 06978, South Korea, kimm@ssu.ac.kr)

If you make a loud voice and lecture with a high voice tone, you usually get a hoarse voice. The voice is generated by ejecting air pressure from the lungs through the folder of the vocal cords to create a tonal tone and resonance in the vocal tract. In this paper, we propose a new loudspeaker amplification tone that induces a hoarse voice when loudly voiced for a long time. In the proposed method, when the hoarseness was eliminated, the average sound pressure level was recovered to +5.2 dB and the spectrum in the high frequency range was improved by the average magnitude level of +12.6 dB. It is confirmed that the clarity of vocalization is more than usual in the measurement of vocal clarity.

9:30

5aBA7. Loudness growth as a function of presentation method: Comparison of normal hearing children with children using cochlear implants. Shubha Tak and Asha Yathiraj (Dept. of Audiol., All India Inst. of Speech and Hearing, Manasagangothri, Mysore, Karnataka 570006, India, shubha_tak@yahoo.co.in)

The loudness of signals is altered in listening devices used by individuals with hearing loss, where soft sounds are made loud and loud sounds made soft. It is speculated that loudness growth would be affected in those using such listening devices. The study aimed to compare the effect of order of intensity presentation on loudness growth in typically developing children and children using monaural cochlear implants. Loudness growth of 3 warble-tones and 3 vowels were examined in 20 typically developing children and 17 children using cochlear implants. The intensity of the stimuli was varied randomly and sequentially. The children rated the loudness of the stimuli on a six-point rating-scale. Only 10 of the 17 children using cochlear implants gave valid loudness growth responses. These 10 children demonstrated no significant difference between the random and the sequential presentation for most of the stimuli. In general, the typically developing children exhibited a significant difference for the extremes in the loudness-growth scale (very soft and very loud). Loudness growth was similar across the methods and participant groups. Thus, a large number of children using cochlear implants are unable to give reliable loudness-growth responses. Those who have a loudness growth perform similar to the typically developing children.

9:45

5aBA8. Egg white-based ultrasound blood mimicking phantom. Yunbo Liu, Brady Connors, Monica Abboud, and Subha Maruvada (FDA, 10903 New Hampshire Ave., WO62RM2126, Silver Spring, MD 20993, yunbo.liu@fda.hhs.gov)

An egg white-based blood mimicking fluid (BMF) was developed and characterized as a blood coagulation surrogate for the acoustical and thermal evaluation of therapeutic ultrasound, especially high intensity focused ultrasound (HIFU) devices. Physical properties, including coagulation temperature, frequency-dependent attenuation, sound speed, thermal conductivity, and thermal diffusivity, were measured as a function of temperature (20–95 ^{o}C). The fluid viscosity was quantified at room and body temperature. With the addition of Nylon particles in the solution, the backscattering coefficient of the BMF was measured and compared before and after a complete thermal coagulation for diagnostic imaging purpose. For a 30 s thermal exposure, the egg white-based BMF (3 mm thickness) started to denature at 65 ^{o}C and coagulate into an elastic gel at 85 ^{o}C. The coagulation temperature can be lowered by adding a small amount of the acid solution to the BMF. HIFU exposure on the BMF was conducted using a flow loop phantom system to simulate and understand the HIFU ablation process and coagulation dynamics. These blood mimicking properties make this egg white-based ultrasound blood phantom a potential tool for pre-clinical bench testing of medical ultrasound devices.

10:00–10:15 Break

10:15

5aBA9. Renal volume reconstruction using free-hand ultrasound scans. Alex Benjamin (Mech. Eng., MIT, 550 Memorial Dr., Apartment 21D-2, Cambridge, MA 02139, arb93@mit.edu), Melinda Chen (Health Sci. and Technol., MIT, Cambridge, MA), Qian Li (Radiology, Massachusetts General Hospital, Boston, MA), Carolina A. Carrascal, Hua Xie (Philips USA, Cambridge, MA), Anthony Samir (Radiology, Massachusetts General Hospital, Boston, MA), and Brian Anthony (Mech. Eng., MIT, Cambridge, MA)

Kidney volume is a predictor of kidney function or the number of surviving nephrons; as such, it is a valuable biomarker in tracking the onset and progression of chronic kidney disease (CKD). Existing techniques to measure renal volumes [CT, MRI, three-dimensional (3D) ultrasound] are limited by high cost, long scan times, and low portability. To overcome these limitations, we propose a real-time, low-cost solution for estimating renal volumes using free-hand ultrasound scans. The method involves an Intel RealSense D415 camera mounted on a conventional one-dimensional ultrasound probe; RGB-D SLAM is used to localize the probe in free space. The acquired two-dimensional images are manually segmented by trained clinicians and combined with their corresponding poses to form a 3D volume. The method was tested on *ex vivo* sheep kidneys embedded in gelatin phantoms. Three different scanning protocols were tested: transverse linear scan (TL), transverse fan scan (TF), and longitudinal fan scan (LF). Ultrasound measured renal volumes were compared to those obtained using the water displacement (WD) method; TL = 66.2 ml, LF = 58.3 ml, TF = 53.3 ml, and WD = 66.6 ml. Further investigation includes automating kidney segmentation, accounting for patient motion, and *in vivo* validation.

10:30

5aBA10. Etiology of the color Doppler twinkling artifact on kidney stones. Scott A. Zinck (Acoust., The Penn State Univ., 201E Appl. Sci. Bldg., University Park, PA 16802, 19scottz@gmail.com)

Rapid Doppler shifts highlight some kidney stones with a rapid change in color in ultrasound imaging which is called the "twinkling artifact." Many hypotheses exist to describe the mechanism of twinkling, including the hypothesis that microbubbles are stabilized in crevices on the surface of kidney stones. The objective is to evaluate the distribution of stable microbubbles on kidney stone surfaces with humidity-controlled scanning electron microscopy (ESEM) and determine whether any sub-surface characteristics as assessed with micro-computed tomography (μCT) may contribute to twinkling. A Verasonics® research ultrasound system with the Verasonics L22-14 and Philips/ATL L7-4 and C5-2 transducers quantified twinkling. Results show a correlation between voids within kidney stones and the locations of twinkling on the stone surface. These voids had a calculated free-field Minneart resonant frequency of 3–9 MHz depending on the size. ESEM revealed that surfaces that twinkled had more crevices with radii corresponding to bubbles with a free-field resonant frequency of 3–7 MHz. These results suggest that internal voids and surface crevices contribute to the twinkling artifact on kidney stones.

10:45

5aBA11. The effect of crystal chemical composition on the color Doppler ultrasound twinkling artifact. Eric Rokni, Scott A. Zinck, and Julianna C. Simon (Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ezr144@psu.edu)

The color Doppler ultrasound twinkling artifact, or localized areas of rapid color shifts, has been used to aid in the diagnosis of kidney stones. Gallstones and some atherosclerotic plaques have also been shown to twinkle, suggesting that crystals, regardless of chemical composition, may harbor the stable microbubbles that are hypothesized to cause twinkling. Here, uric acid, cholesterol, and calcium phosphate crystals, which are often present in sites of pathological biomineralization, were grown in the lab using previously developed techniques and imaged with the Verasonics® research ultrasound system using Philips/ATL P4-2 and Verasonics L22-14 transducers. The magnitude of the Doppler signals was quantified and compared amongst the three crystal chemical compositions. Preliminary results show that all three crystal types displayed twinkling with the strongest magnitude appearing in the cholesterol crystals. These results suggest that twinkling is

very sensitive in the presence of crystals regardless of the chemical composition. Future work will explore the use of histotripsy to mechanically fractionate crystals as a novel treatment for pathological biomineralization.

11:00

5aBA12. Introducing Feelix, a digital stethoscope incorporating active noise control and automatic detection of lung sound abnormalities. Ellington West, Ian McLane (Sonavi Labs, 1100 Wicomico Rd., Ste. 600, Baltimore, MD 21230, ellington@sonavilabs.com), Daniel McLane (Phase Margin, Chatsworth, CA), Dimitra Emmanouilidou (Microsoft Res., Seattle, WA), Mounya Elhilali, James E. West (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Arthur Ward, Ilene Busch-Vishniac (Sonavi Labs, Baltimore, MD), Jaishree McLane (Sonavi Labs, Chatsworth, California), and Brandon Dottin-Haley (Sonavi Labs, Baltimore, MD)

Pneumonia is the top disease killer of children under the age of 5 worldwide, and while the definitive diagnosis of pneumonia typically is made with x-rays, only 5% of the world's population has ready access to radiographic equipment. To address this problem, Sonavi Labs has introduced the Feelix line, a line of digital stethoscopes focused on abnormal lung sound detection. Compared to traditional, mechanical stethoscopes, the Feelix line offers three advantages: reduced sensitivity to placement of the sensing head, active noise cancellation, and automatic detection of lung sound abnormalities using an algorithm that incorporates machine learning. This new line of digital stethoscopes is appropriate for under-resourced regions of the world, for noisy clinics, and for medical professionals who wish to improve their ability to hear lung sounds. Using this line of digital stethoscopes is preferable to alternative methods that are under study, such as lung ultrasound, as its use is entirely noninvasive and its cost is a fraction of that of x-ray machines or ultrasound imagers.

11:15

5aBA13. Ultrasonography education in the first medical-engineering based college. Olivia Coiado (Carle Illinois College of Medicine, Univ. of Illinois at Urbana-Champaign, 1914 Max Run Dr., Champaign, IL 61822-3449, oliviacoiado@hotmail.com)

This work presents an ultrasonography laboratory practice for the first year medical students of Carle Illinois College of Medicine. Carle Illinois is the first college of medicine in the nation designed at the intersection of engineering and medicine. The curriculum, which integrates basic and clinical sciences with engineering and innovation, includes activities for students to navigate through a guided ultrasound practice. In the individual session, students were introduced to an ultrasound machine (Sonosite Edge, FUJIFILM SonoSite, Inc., Bothell, WA) and discussed the ultrasound principles. As preparation for the session, students received a guide where they answered a few questions. The guided practice consisted of describing the parts of a basic ultrasound machine. Students explained ultrasound principles, such as the relationship between frequency and wavelength, different ultrasound imaging modes and types of ultrasound transducers, frequency of ultrasound transducer for diagnosis and therapy, and the continuous Doppler mode. In the end, students had the opportunity to scan the carotid artery and analyze the ultrasound principles discussed. After the guided session, students were allowed to perform scans on their own. The students appreciated the hand-on experiences gained by participation in the ultrasonography-guided session.

FRIDAY MORNING, 17 MAY 2019 JONES, 8:15 A.M. TO 12:00 NOON

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics II

Kyle S. Spratt, Cochair
Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713

Kevin M. Lee, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Contributed Papers

8:15

5aPA1. Sound speed measurements in Kentucky bourbon whiskey. Stanley A. Cheyne (Dept. of Phys. & Astronomy, Hampden-Sydney College, Hampden-Sydney, VA 23943, scheyne@hsc.edu)

Kentucky is home to most of the bourbon whiskeys sold world wide. Sound speed measurements were made in a number of Kentucky bourbon whiskeys. Results of a variety of whiskeys, some having different alcohol content,s using a time-of-flight technique will be presented. In addition, phase speed measurements in impedance tubes using a transfer function technique will be presented. Experimental results will be compared to what is expected in a water/ethanol mixture.

8:30

5aPA2. Acoustic and dynamic characterization of wooden baseball bats. Kenneth A. Pestka II and Derek E. Holmberg (Chemistry and Phys., Longwood Univ., 201 High St., Farmville, VA 23909, pestkaka@longwood.edu)

We will present the results of several experiments used to characterize the properties and performance of multiple baseball bats composed of four different species of wood including soft maple, hard maple, birch, and ash. Resonant ultrasound spectroscopy was used in an effort to obtain the complete orthotropic tensor of the different wood species. In addition, ultrasonic pulse-echo measurements and dynamic impulse time-of-flight

measurements were performed to obtain longitudinal sound speeds, which in combination with bat geometry, mass, and density enabled the determination of the longitudinal Young's moduli. Bat performance was then experimentally determined via a newly developed non-destructive impact method. The physical acoustic parameters that most significantly correlated with optimum bat performance will also be presented.

8:45

5aPA3. Spontaneous thermokinetic amplitude modulation in the "burning wire" lecture demonstration. Steven L. Garrett (Laboratoire d'Acoustique, Le Mans Université, 151 Sycamore Dr., State College, PA 16801, sxg185@psu.edu) and Guillaume Penelet (Laboratoire d'Acoustique, Le Mans Université, Le Mans, France)

A thin nichrome wire that is tensioned by a mass and driven by an alternating electrical current, with a magnet placed along the wire's length, provides a popular lecture demonstration. Due to the convective cooling of the portions of the wire moving with the greatest velocity, in a darkened room, only the regions near the velocity nodes will be visible due to the orange glow of the heated wire. The thermal expansion coefficient of the wire forces the tensioning mass to change its height as temperature changes. By adiabatic invariance, the changes in the mass's elevation cause the natural frequency to be shifted. Competition between the thermal inertia of the wire and the convective heat transfer coefficient introduces an exponential thermal relaxation time so that the amplitude of vibration is dependent on the ratio of the constant drive frequency to the changing resonance frequency at an earlier (retarded) time. These thermal and kinetic effects are incorporated into three coupled nonlinear ordinary differential equations that are separated by the method of multiple time scales and are solved numerically, reproducing both the spontaneous appearance of stable periodic amplitude modulation and the hysteretic behavior observed with the increasing or decreasing drive frequency.

9:00

5aPA4. Performance analysis of a looped travelling-wave thermoacoustic engine with phase-change. Rui Yang, Avshalom Offner, Avishai Meir, and Guy Z. Ramon (Nancy and Stephen Grand Technion Energy Program and Dept. of Civil & Environ. Eng., Faculty of Civil and Environ. Eng., Technion-Israel Inst. of Technol. Haifa 3200003, Israel, rui.yang@campus.technion.ac.il)

Traditional thermoacoustic engines using conduction-driven, sensible heat transfer are unable to utilize low-grade thermal energy efficiently. Recently, it has been demonstrated that a phase-change thermoacoustic engine can initiate oscillations at a very low temperature difference; however, the steady-stage performance has yet to be extensively studied. In this work, a phase-change thermoacoustic engine with a looped resonator was simulated, based on a linear phase-change thermoacoustic theory. A binary gas mixture consisting of an inert gas and a condensable component was adopted as the working fluid. We numerically investigated the distributions of pressure, volumetric velocity, and acoustic power in the system. We also examined the steady-state performance (characterized by the pressure amplitude, acoustic power, and thermal efficiency) of the system under different mean pressures and temperature differences. The results show that the phase-change thermoacoustic engine can be driven by very low temperature differences (< 50 K) much more efficiently than its dry equivalent (i.e., the same system but without the condensable component). The findings demonstrate the promising potential of generating acoustic power through low-grade heat recovery, which can then be converted into electricity and cooling.

9:15

5aPA5. Measuring the transmission coefficient of a steel plate using an angular spectrum approach. Kyle S. Spratt, Benjamin C. Treweek, Kevin M. Lee, Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78758, sprattkyle@gmail.com)

Ultrasonic measurements were performed on a thin steel plate submerged in water in order to obtain the plane-wave transmission coefficient as a function of angle and frequency. An angular spectrum approach was utilized wherein the acoustic field parallel to the plate was scanned with sufficient spatial accuracy to obtain a frequency-wavenumber description of the transmitted field, representing an expansion of the field into time-harmonic plane waves propagating within a range of angles. Isotropy of the plate material and symmetry of the measurement configuration allowed for a vast reduction of the number of scan points required in order to obtain the transmitted angular spectrum. Once the transmitted acoustic data is acquired, the specific method of windowing the data in space and time can have a drastic effect on the resulting angular spectrum. With sufficient care, a broadband estimation of the transmission coefficient with a high angular resolution is obtained. Finally, inverse methods were utilized in order to estimate the thickness and material properties of the plate using the measured transmission coefficient data. [Work supported by ONR.]

9:30

5aPA6. Generating ultrasonic acoustic holograms with circular arrays. Paul T. Johnson (Phys., Illinois Wesleyan Univ., 201 E Beecher St., Bloomington, IL 61701, pjohnso3@iwu.edu)

Acoustic holograms are formed by interfering multiple sonic sources of identical frequency and differing phase. By using ultrasonic transducers, three-dimensional printed arrays, and an iterative approximation algorithm, it is possible to computer generate holographic arrays at 40 kHz in different configurations. By using circular arrays, I am able to explore vortex generation in an ultrasonic with a cost-effective desktop setup. These holograms can be measured and plotted using another transceiver. I also am able to make predictions about how to best optimize the designs for a circular array. This research opens up questions to be explored with even more powerful setups.

9:45

5aPA7. Photoacoustic harmony: Using musical concepts to study mixed gases through PA spectroscopy. Trevor Paratore and Han Jung Park (Chemistry, Univ. of Tennessee at Chattanooga, 615 McCallie Ave., Chattanooga, TN 37403, hanjung-park@utc.edu)

Photoacoustic Spectroscopy was applied to study the intensity of musical notes in harmony created by the photoacoustic signals of ethylene (C_2H_4) and sulfur hexafluoride (SF_6) gases. Two CO_2 lasers with a wavelength of 10.6 μm were modulated using two separate mechanical choppers. It was observed that musical notes with a frequency approximately that of the fundamental frequency of the resonator had higher intensity peaks than those further away. To equilibrate the intensities of these notes for fuller tones in the musical chords of interest, a second resonator, sulfur hexafluoride, having vastly different resonance properties from ethylene, was added to the gas cell. It was observed that this only moderately affected the outcome for the overall tone of the chords of interest. However, as expected, sulfur hexafluoride offered a more dynamic range of tones due to its large variance in resonance frequency with concentration.

10:00–10:15 Break

10:15

5aPA8. High temperature dependent elastic properties of single crystalline SnSe. Ashoka Karunarathne, Josh R. Gladden (Phys. and Astronomy & NCPA, Univ. of Mississippi, 145 Hill Dr., University, MS 38677, atholang@go.olemiss.edu), and Gautam Priyadarshan (National Ctr. for Physical Acoust., Univ. of Mississippi, University, MS)

In recent years, single crystalline SnSe has been widely studied as a thermoelectric material which exhibits a high thermoelectric figure of merit ($ZT \sim 2.6$ at $T > 900$ K). The efficiency has been enhanced due to the significantly low thermal conductivity, arising from the strong lattice anharmonicity of the layered orthorhombic crystal structure. A study of the thermoelastic properties of single crystalline SnSe is essential to gain a better understanding of the lattice anharmonicity and the low thermal conductivity. The highly anisotropic orthorhombic crystal structure of SnSe results in nine independent elastic constants. This work seeks to investigate the temperature dependent elastic constants of single crystalline SnSe, using

resonant ultrasound spectroscopic (RUS). Here, we report the experimentally observed elastic constants and their variation at elevated temperature from room temperature to ~775 K. The temperature dependent elastic and thermal properties have been derived from the elastic constants and compared to independent heat capacity data.

10:30

5aPA9. High temperature resonance ultrasound spectroscopy studies of single crystal minerals. Sumudu P. Tennakoon and Mainak Mookherjee (Earth, Ocean and Atmospheric Sci., Florida State Univ., Tallahassee, FL 32306, stennakoon@fsu.edu)

Elasticity of mineral phases at temperature and pressure conditions relevant to earth interior is essential for constraining the composition and the dynamics of the deep Earth. To constrain the temperature dependence of elasticity of hydrous minerals relevant to the earth's subduction zones, we used high temperature Resonance Ultrasound Spectroscopy (RUS). We explored the temperature dependence of the full elastic moduli tensor, speed of sound, attenuation, and anisotropy of naturally occurring single crystal topaz ($Al_2SiO_4F_{1.42}(OH)_{0.58}$) [1] with a rectangular parallelepiped geometry. The RUS spectra are influenced by the geometry, density, and the full elastic moduli tensor [2]. We determined the crystal symmetry and crystallographic alignment of the crystals using X-ray diffraction. The geometry and density are well constrained from previous results on thermal expansion. The elasticity results on topaz are in good agreement with previous studies. We combine the high temperature elasticity of natural topaz with the results from the *first principles* on end member hydrous topaz and shed insight into how pressure temperature and composition, i.e., the fluorine and hydrogen content could influence elasticity of minerals in the Earth's interior, in particular in subduction zone settings. [Work supported by U.S. NSF Award Nos. EAR-1634422 and EAR-1753125.] References: [1] Tennakoon *et al*., *Sci. Rep*., 8, 1372 (2018). [2] A. Migliori and J. D. Maynard, *Rev. Sci. Instrum*.76, 121301 (2005).

10:45

5aPA10. Nonlinear resonant ultrasound spectroscopy using electromagnetic excitation. Joshua F. Gregg and Brian E. Anderson (BYU Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, joshygregg1@gmail.com)

Nonlinear resonant ultrasound spectroscopy (NRUS) is a method that can be used for detecting microcracks in structures. NRUS detects shifts in resonance frequencies versus excitation amplitude. Excitation of a sample is typically done with a piezoelectric transducer. Here, the application of an electromagnetic excitation is explored as an alternative for NRUS excitation. It involves gluing a coil of wire onto the end of a rod sample and placing it in a magnetic field. By controlling which part of the coil is in the strongest part of the magnetic field, the principle direction of the driven oscillations in the rod can be controlled. This method allows selective excitation of longitudinal, torsional, and bending vibrations, and the measurement of the nonlinear properties of the sample for each type of vibration is therefore possible. Both a piezoelectric transducer and electromagnetic excitation are used to measure the nonlinear elastic parameters of a Berea sandstone rod. The effects of heating of the sample and slow dynamics are explored and compared with each type of excitation. The purpose of this presentation is to illustrate the usefulness of the electromagnetic method and outline its differences from the piezoelectric method.

11:00

5aPA11. Temperature-dependent studies of high-performance foamed cements. Eric S. Davis, Hung Doan, Blake Sturtevant, and Cristian Pantea (Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, esdavis@lanl.gov)

In this study, through-transmission acoustic time-of-flight experiments were used to study the variation of the mechanical properties of foamed cements when subjected to realistic field temperatures. Since temperature increases by ~25 °C/km in depth, it is important to understand the properties of foamed cements used in wellbore applications at relevant depths. By studying these foamed cements up to around 165 °C, characterization up to ~6 km in depth is achieved, which covers almost all drilling depths. In particular, it was found that foamed cements containing a higher percentage of air were less affected by temperature, and that Young's modulus and the compressional sound speed do not decrease linearly with the increasing air content, as one might expect. The sound speeds varied from 2942 m/s for 10% foamed cement, to 2500 m/s for 30% foamed cement at room temperature. Young's modulus varied from 16.0 GPa for 10% foamed cement to 9.2 GPa for 30% foamed cement at room temperature. The softening rate with heating also decreased with the increasing air content, with 10% foamed cement decreasing by 7.8% in Young's modulus with heating from room temperature to ~165 °C, while 30% foamed cement only decreased by 5.5%.

11:15

5aPA12. Acoustics-based 3-dimensional temperature gradient determination in paraffin wax. Cristian Pantea (MPA-11, Los Alamos National Lab., MS D429, Los Alamos, NM 87545, pantea@lanl.gov), John Greenhall (MPA-11, Los Alamos National Lab., Santa Fe, NM), Eric S. Davis, and Craig Chavez (MPA-11, Los Alamos National Lab., Los Alamos, NM)

There are currently no technologies to noninvasively assess explosive condition from outside its external confinement for explosive safety activities. Knowledge of the explosive condition inside an item can help to estimate the time that remains before thermal damage ignites the explosive, or how the explosive responds to various actions. Acoustic technologies based on COTS hardware and designed to determine temperature gradients have been in development for several years with no demonstrated success. Conventional acoustical sources at low frequencies tend to have a large beamspread with side lobes, resulting in significant acoustical energy losses and very poor spatial resolution. The use of a unique collimated low-frequency sound beam and beam scanning, developed recently in our lab, was investigated in paraffin wax for acoustic temperature determination. Acoustic through-transmission experiments, and knowledge of the relationship between sound speed and temperature led to spatially resolved three-dimensional temperature field inside the sample, in good agreement with those measured using thermocouples.

11:30

5aPA13. A comparative study of the dynamics of laser and acoustically generated bubbles in viscoelastic media. Jonathan R. Sukovich (Biomedical Eng., Univ. of Michigan, 1410 Traver Rd., Ann Arbor, MI 48105, jsukes@umich.edu), Chad Wilson (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Timothy L. Hall (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Lauren Mancia, Mauro Rodriguez, Eric Johnsen (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), and Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Here, we present results from experiments comparing the first-cycle growth and collapse dynamics of acoustically and laser-nucleated single bubbles in water and agarose gels of varying stiffness. Experiments were carried out in a custom-built spherical vessel which allowed both acoustically and laser-nucleated bubbles to be generated at its center. Bubbles were nucleated in water and agarose gels with stiffnesses ranging from 1.13 kPa to 570 kPa. Acoustically nucleated bubbles were generated using pulses delivered from 1 MHz transducer elements mounted in the vessel with pulse durations of <2 acoustic cycles. Laser-nucleated bubbles were generated using a pulsed Nd:YAG laser focused to the center of the vessel through an embedded window. Bubble dynamics were measured via optical imaging. The maximum radii of generated bubbles decreased with the increasing material stiffness for bubbles nucleated by both mechanisms. The growths and collapses of the bubbles occurred symmetrically (in time) about the maximum radius in water but not in gels, where the duration of growth decreased more rapidly than collapse as gel stiffness increased. Nucleation-mechanism-dependent differences in the growth-collapse asymmetry were observed which indicated that the growth duration of the laser-nucleated bubbles was longer than the acoustically nucleated bubbles in water and high water-content gels.

11:45

5aPA14. The effect of tapered entrance in nozzle flow cavitation. Grace McDonough, Jennifer M. Moreira, Ankush Gupta, Emily Ryan, Sheryl Grace, and R. Glynn Holt (Boston Univ., 110 Cummington Mall, B-08, Boston, MA 02215, gracemcd@bu.edu)

Cavitation erosion is a common cause of damage of fuel injectors, pumps, and other hydrodynamic equipment. For fuel injectors, cavitation can also have a beneficial effect on the resultant spray angle and aerosol size distribution. Because collapsing cavitation bubbles emit pressure waves, the occurrence and extent of cavitation can be characterized using an ultrasonic transducer in the spirit of passive cavitation detection, commonly employed in biomedical acoustics. Experiments use a two-part acrylic nozzle: a contraction portion of length 1 in. (contracting from the feed tube diameter of 1.14 in. to the diameter of the cylindrical portion) and a cylindrical outlet of length 1 in. Two different outlet diameters have been considered. Water is caused to flow at increasing rates which span the inception and development of hydrodynamic cavitation. Acoustic and optical results will be presented and compared. The results indicate that introducing a tapered change in the nozzle diameter delays the onset of cavitation with respect to the mean flow rate, as does decreasing the taper angle with respect to the flow axis. [Work supported by DOE DE-EE0007332.]

FRIDAY MORNING, 17 MAY 2019 GRAND BALLROOM C, 8:00 A.M. TO 12:00 NOON

Session 5aSC

Speech Communication: Production (Poster Session)

Olga Dmitrieva, Chair
Purdue University, 640 Oval Drive, Stanley Coulter 166, West Lafayette, IN 47907

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

Contributed Papers

5aSC1. Patterns of vowel nasality in Brazilian Portuguese. Luciana Marques (English, Colorado State Univ., 359 Willard O. Eddy Hall, 1773 Campus Delivery, Fort Collins, CO 80523-1773, luciana.marques@colostate.edu) and Rebecca Scarborough (Linguist., Univ. of Colorado at Boulder, Boulder, CO)

This study aims to establish differences in patterns of acoustic nasality between nasal and nasalized vowels in Brazilian Portuguese (BP). The difference in how nasality is implemented in the acoustic signal is hypothesized to reflect the different phonological status of these two vowel categories. Acoustic nasality is the difference between the amplitude of the first formant (A1) and the amplitude of a harmonic in the low region of the vowel spectrum (P0) (Chen, 1997). The smaller the difference between these peaks, the more nasal a vowel is. A1-P0 measurements were taken at different time points from recordings of oral, nasal, and nasalized vowels in BP of five qualities. Results show that nasal and nasalized vowels were similar in all qualities. However, the way nasality is implemented over time between the vowel categories are different, with acoustic nasality in nasal vowels increasing more rapidly than in the nasalized counterparts. Although the patterns of nasality found are not consistent with the expected nasal profile difference (Cohn, 1990), the difference in acoustic nasality between nasal and nasalized vowels is consistent enough to consider it result of how the feature [nasal] is implemented in the language due to its variable phonological status in BP.

5aSC2. The prosody of corrective focus and its domain in Korean. Miran Oh and Dani Byrd (Linguist., Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, miranoh@usc.edu)

Speakers place focus on specific linguistic elements using systematic modulations to highlight new or important information. Focus effects have primarily been studied for domains larger than a segment, such as syllables or words. By eliciting focus on segment-sized units using a corrective focus task, this study examines at what granularity or scope speakers manifest focus and its acoustic consequences for consonants differing in gestural and tonal compositions. Data were obtained from 16 native Korean speakers producing corrective focus on segments having varying syllable-internal positions (i.e., onset, nucleus, versus coda focus) and varying segmental compositions (i.e., tense and lax consonants). The results indicate that the patterns of focus modulation do not differ depending the on anticipated sub-syllabic focus position but rather are exhibited through the syllable. This suggests that the domain of focus extends beyond segment-sized intervals. Moreover, focus is manifested with systematic variations as a function of the gestural compositions active during its domain. Finally, there is some support for an interaction among multiple prosodic gestures—focus gestures and accentual phrase tone gestures. Overall, evidence suggests that focus has a syllable-sized (or larger) scope of effect that is sensitive to co-active articulatory and prosodic (tonal) gestures. [Work supported by NIH.]

5aSC3. A cross-linguistic comparison on the use of acoustic cues for ambiguity resolution. Hyunah Baek (Dept. of Linguist., Stony Brook Univ., Stony Brook, NY 11790, hyunah.baek@stonybrook.edu)

In producing structurally ambiguous sentences, speakers often rely on prosodic acoustic cues (duration, pitch, and intensity) to enhance communication. This study compared speakers of two languages in their use of these cues for ambiguity resolution. Ten English and 10 Korean speakers completed a production task in their native language. Stimulus sentences included a relative clause which could modify either the first or the second noun (e.g., *the boss of the clerk who was dishonest*). The participants produced each sentence twice, being directed each time to convey one or the other of the two possible meanings. English speakers used higher pitch and greater intensity on a word when it was being modified, but their use of durational cues was less consistent. Korean speakers, in contrast, consistently used lengthening on phrase-final word

and longer pauses at the major constituent break, but their use of pitch and intensity was not consistent. These results suggest that to resolve parallel structural ambiguities, English speakers use pitch and intensity cues while Korean speakers use durational cues. I argue that this difference relates to the prosodic systems of each language, with English using pitch accents associated with words and Korean using boundary tones associated with accentual phrases.

5aSC4. Coarticulation varies by consonant identity in adult and child speech. Margaret Cychosz (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720, mcychosz@berkeley.edu)

Consonant-vowel coarticulation varies by manner of articulation: speakers coarticulate less between segments in V[stop] sequences than V[glide]. There are different approaches to modelling consonant identity effects upon coarticulation (Iskarous *et al*., 2013). However, these effects have not been studied in non-literate populations—young children or non-literate adults. This is relevant since literacy heightens phoneme awareness, affecting coarticulation. 40 South Bolivian Quechua speakers who were not literate in the language, 30 children (aged 5–10) and 10 adults, completed a picture-prompted word elicitation task. Coarticulation was measured between [a] and the consonants /p, t, k, k^h, q^h, ch, s, m, w/ in VC and CV syllables. Because the child speech apparatus poses problems for typical spectral analysis, we quantify coarticulation as the Euclidean distance between Mel frequency cepstral coefficient vectors averaged over adjacent phones (Gerosa *et al*., 2006). Mixed effects models predicting the Euclidean distance between adjacent phones show that all age groups are sensitive to changes in consonant manner. Adults and children restrained coarticulation more between sequences containing obstruents than glides or liquids. These results suggest that literacy and heightened phoneme awareness are not required for consonant identity to affect coarticulation.

5aSC5. The acoustic effects of emphatic and pharyngeal consonants on adjacent vowels. Laura R. Faircloth (Dept. of Linguist., Univ. of Texas at Austin, 305 E 23rd St., STOP B5100 RLP 4.304, Austin, TX 78712, lrfaircloth@utexas.edu)

Emphatic consonants in Arabic are coronal obstruents with a debated secondary articulation, contrasting with plain coronals and affecting adjacent vowels (Watson, 2007). The secondary articulation has been claimed to be pharyngeal, so pharyngeals may affect vowels similarly. Eight speakers of Palestinian Arabic recorded words with /a:/, /i:/, or /u:/ following plain /s/, empathic /$s^ʕ$/, or pharyngeal /ħ/. The effects of plain and emphatic consonants on /a:/ have been studied, but other vowels may be affected differently. The CoG of /$s^ʕ$/ was lower than /s/, differing from previous studies (Card, 1983). F1 and F2 measurements from the onset, midpoint, and offset of the vowel showed the extent of emphatic effects. F2 in /a:/ and /i:/ and F1 in /a:/ and /u:/ adjacent to /$s^ʕ$/ were lower than adjacent to /s/ or /ħ/. Both suggest a backed and raised tongue in emphatics. F1 was only higher adjacent to /ħ/ at the onset, while emphatics affected the entire time course. Pharyngeal /ħ/ caused typical coarticulation, while the longer effects of emphatic /$s^ʕ$/ suggest a phonological process. The differences between emphatic and pharyngeal consonants also suggest that emphatic consonants in this dialect are produced with uvular constriction, not pharyngeal as has been claimed.

5aSC6. Trade-offs between labial and lingual activity in the Suzhou Chinese labial fricative vowels. Matthew Faytak, Jennifer Kuo, and Shunjie Wang (Dept. of Linguist., Univ. of California, Los Angeles, UCLA, 3125 Campbell Hall Box 951543, Los Angeles, CA 90095, faytak@ucla.edu)

We examine trading relations between lingual and labial activity for three back vowels with different labial constrictions in Suzhou Chinese. Specifically, we compare the labial "fricative vowels" [ɯ͡β] and [ɯ͡v], which exhibit a compressed lip opening and a labiodental constriction, respectively, with rounded [u]. Linear mixed-effects modeling of indices of tongue contour shape extracted from ultrasound video demonstrates that the degree of tongue dorsum raising differs systematically from vowel to vowel: [ɯ͡β] and [ɯ͡v], which have more labial constriction than [u], exhibit reduced back raising activity compared to [u]. We additionally report on correlations between lingual activity and labial activity based on the frontal lip shape and lip aperture data collected from a time-aligned video channel.

5aSC7. Structured variation in Suzhou Chinese fricative consonants and vowels. Matthew Faytak (Dept. of Linguist., Univ. of California, Los Angeles, UCLA, 3125 Campbell Hall Box 951543, Los Angeles, CA 90095, faytak@ucla.edu)

Uniform phonetic implementation of phonological primitives has been suggested to be a low-level constraint on speech production. Suzhou Chinese offers an example of how this constraint may shape the development of systems of phonological contrast: a pair of recently innovated "fricative vowels," rounded and unrounded, have come to be produced with the addition of noticeable fricative noise. This noise is spectrally similar to that of a fricative consonant, most frequently alveolopalatal /ɕ/ but occasionally /s/, suggesting that speakers have begun to refer to shared acoustic targets for these phonotactically dissimilar sounds. In order to assess the degree of acoustic similarity among these segments, we compare the fricative noise in the unrounded and rounded fricative vowels of 43 speakers of Suzhou Chinese with the fricative noise of each speaker's onset fricative consonants /ɕ/ and /s/. Spectral parameters analyzed are chosen to reflect constriction anteriority in strident fricatives. Correlations in these parameters as implemented across consonantal onsets such as /ɕ/ and /s/, and the unrounded fricative vowel and the rounded fricatie vowel are examined.

5aSC8. A kinematic investigation of coarticulation resistance in Modern Greek. Evdoxia Doli (Behavioral and Brain Sci., Callier Ctr. for Commun. Disord., The Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75207, exd160330@utdallas.edu) and William F. Katz (Behavioral and Brain Sci., The Univ. of Texas at Dallas, TX)

In order to examine the coarticulation resistance hypothesis (CRH) this study investigated patterns of CV and C1C2V overlap in consonant clusters and matching singletons in Greek, including five places of articulation in C2V position (bilabials, labiodentals, interdentals, alveolars, and velars). Stimuli were recorded from three talkers producing eight repetitions of each item, embedded in a carrier phrase. Kinematic and acoustic data were acquired with an electromagnetic articulography (EMA) system. Temporal measures were obtained by determining timing lags for both singletons and clusters, while spatial measures were found by calculating the Euclidean Distance (ED) of the tongue back sensor from the C2V velocity peak to that of the following vowel, for both singletons and clusters. Logarithmic ED ratios for each stimulus triplet (e.g., "ba"-"la"-"bla") were then computed to test whether C2V allows C1V to exert coarticulatory influence on the vowel. Preliminary findings from both the temporal and spatial analyses suggest pronounced differences in complex versus simple syllable organization as a function of C2V place of articulation. The extent to which these results support the CRH will be further discussed.

5aSC9. Acoustical emphasis not possible: The case of emphatic morphemes in Lakota. Brooke L. Kidner (Linguist., Univ. of Southern California, 620 McCarthy Way, #273, Los Angeles, CA 90089, bkidner@usc.edu)

In languages like English, emphasis can be expressed acoustically by an increase in pitch or amplitude. These types of pitch excursions and increase in amplitude are often how metrical stress is marked in languages. This type of representation of metrical stress is true for Lakota. While this type of acoustic representation of stress is not uncommon, it does present a unique problem for Lakota speakers when it comes to encoding emphasis acoustically in their speech. Lakota (a member of the Dakota family of languages with less than 9000 speakers), has a very rigid set of rules for where stress can be applied in languages, often having several minimal pairs that differ only in which syllable receives primary stress. As such, they are unable to utilize these common acoustic markers for emphasis in their speech. Field research conducted on the Pine Ridge Reservation (2018, 2016) has resulted in the discovery that Lakota possesses two emphatic morphemes: [kʃto] for female speakers and [jelo] for males, which speakers utilize to encode emphasis into their speech.

5aSC10. The lingual topography of American English laterals in onsets and codas. Kenneth de Jong, Kelly Berkson (Dept. of Linguist., Linguist, Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu), Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), Samantha Myers, and Amanda Bohnert (Linguist., Indiana Univ., Bloomington, IN)

This paper reports on a three-dimensional/four-dimensional ultrasound investigation of American English laterals. Traditional descriptions of English distinguish between pre-vocalic (onset) laterals ('light-l'), said to involve a coronal gesture creating an alveolar occlusion on the center line with lateral venting, and post-vocalic (coda) laterals ('dark-l'), said to involve both the coronal gesture and an additional dorsal constriction gesture. Recent work suggests that this view of laterals may be overly simple: e.g., onset laterals in American English may also have a dorsal gesture [Rhodes *et al*., *JASA*137(4), 2268–2269 (2015)]. Previous work also noted a persistent tongue configuration involving deep cupping of the midline in the palatal region yielding a raised tongue tip and dorsum around the cupped region. This configuration appears in both onsets and codas, even for a speaker who exhibited no coronal contact [Berkson *et al*., *ICASSP* (2017), pp. 5080–5084]. This study presents configurations for 20 college-aged native speakers of English (10 female) who show individual differences in (i) the presence of palatal grooving and (ii) consistency between onset and coda configurations. We contend that the onset versus coda lateral distinction cannot be reduced to a single description in terms of lingual configuration.

5aSC11. Acoustic and articulatory investigation of the Mauritian vowel inventory. Samantha Myers (Lingustics, Indiana Univ., Bloomington, IN), Fabiola Henri (Linguist., Univ. of Kentucky, Lexington, KY), and Kelly Berkson (Dept. of Linguist., Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

While there have been a decent amount of morphosyntactic and sociolinguistic studies on Mauritian, a "French-based" creole spoken by most of the population of Mauritius (e.g., Baker, 1972; Alleesaib, 2012; Henri, 2010; Miller, 2015; Syea, 2013), phonetic or phonological description on the language hardly exists. Pudaruth (1993) proposes a phoneme inventory consisting of 19 consonants and 8 vowels, intuitively noting that rhotics preceding vowels are weakened, almost unpronounced. In the present work, we empirically investigate the vowel inventory of Mauritian using acoustic and articulatory data. In addition to presenting basic acoustic measures (e.g., vowel spaces), we note the presence of a rhotic, and report acoustic measures which suggest strong variation between the orthographic "a" that precedes "r" and that which precedes all other written consonants. Ultrasound imaging confirms that the pre-rhotic "a" differs from other "a"s articulatorily.

5aSC12. Quantity, quality, both or neither? Vowel contrasts in Hakha Chin monophthongs. Rebecca Haley, Samson A. Lotven, James Wamsley, and Kelly Berkson (Dept. of Linguist., Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

Hakha Chin—a Tibeto-Burman language spoken in Chin State in western Myanmar—is reported to have phonemic length distinctions in its monophthongs (Melnik, 1997; Peterson, 2003; Maddieson, 2004) (e.g., [ii] vs. [i], [aa] vs. [a]). These pairs are also sometimes represented with tense/lax vowels (e.g., [i] vs. [ɪ], [a] vs. [ə]) (Peterson, 2017). Indeed, the results of a perception study conducted by Mortenson and Van Bik (2002) found variable response patterns in different dialect groups. Listeners from Hakha, the city after which the language is named, relied heavily on spectral cues related to quality to distinguish between long and short vowels, while listeners from the town of Thanthlang relied mainly on durational cues to perform the same task. In the current study, we present the first instrumental acoustic analysis of Hakha Chin vowel quality in order to aid in elucidating the nature of the contrast. Data from three college-aged native speakers (two male) suggest that neither duration nor quality play key roles in distinguishing between the so-called long and short vowels of Hakha Chin. Rather, as originally suggested by Maddieson (2004), the length associated with long vowels is mainly realized through lengthening of sonorant codas.

5aSC13. Pitch accent and the three-way laryngeal contrast in North Kyungsang Korean. Young Hwang, Samson A. Lotven, and Kelly Berkson (Dept. of Linguist., Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

Previous description of Korean's three-way voiceless stop contrast notes differentiation in both voice onset time (VOT) and fundamental frequency (f0) on following vowels (Cho *et al*., 2002; Kang and Han, 2013; Kang, 2014). In a sound-change in-progress, however, VOT for lenis and aspirated stops has merged. The two are now differentiated primarily through f0 differences (Kang and Guion, 2008; Lee *et al*., 2013; Silva, 2006). North and South Kyungsang (NK and SK) are pitch accent dialects. SK has not undergone the change (Lee and Jongman, 2012): VOT is still a primary cue, potentially because f0 is already used for the pitch accent system (Lee *et al*., 2013). In NK, meanwhile, VOT for lenis and aspirated stops overlaps more for female than for male speakers, suggesting a possible change-in-progress (Holliday and Kong, 2011). As questions about the robustness of this change remain, we report data from 23 native NK speakers (6 older and 6 younger males; 5 older and 6 younger females). VOT and f0 findings suggest that the new pattern is incipient in NK: Older male talkers show the old pattern, younger female talkers the new pattern, and older female and younger male talkers fall between the two extremes.

5aSC14. A phonetic study of Swahili voiced stops. Jeremy Coburn and Nils Hjortnaes (Linguist., Indiana Univ., Ballantine Hall 821, 1020 E. Kirkwood Ave., Bloomington, IN 47405, nhjortn@iu.edu)

Despite a rich history of general linguistic research, discrepancy exists within the literature on the phonetic description of voiced stops in Swahili. Polome (1967) reports the voiced stops are phonemically implosive, with plosive allophones in some environments, but which may be strictly egressive for some speakers. Additionally, he states that other speakers have implosives for all stops (i.e., [ɓ], [ɗ], [ɟ]) except the voiced velar plosive [g]. Contemporary descriptions classify all the voiced stops as pulmonic voiced plosives (Mohamed, 2001). This study is an acoustical examination of voiced stops of Swahili speakers from various regions of Tanzania and Kenya to determine the presence of implosives in Swahili and its dialects. The parameters under observation include: (1) voicing onset time (VOT), (2) voicing terminating time (VTT), (3) amplitude, and (4) closure duration as described in several works (Nihalani, 1986; Clements and Osu, 2002; Chávez-Péon, 2005; especially, Cun, 2009). This study discusses preliminary findings on the analysis of five speakers' productions. Four speakers were found to have implosives in all cases except prenasalized voiced stops. One speaker produced plosives in all instances. The results suggest that variation does occur and further research is needed to establish a better description of the language.

5aSC15. Effect of simultaneous lip-tube and auditory feedback perturbations. Lambert Beaudry (Linguist., Universite du PQ a Montreal, Montreal, QC, Canada), Pascal Perrier (GIPSA-Lab, Saint Martin d'Hères, France), and Lucie Menard (Linguist., Universite du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

In a series of previous experiments, it has been shown that when required to produce the back rounded vowel /u/ with a lip-tube perturbation that prevents lip rounding, speakers reached the auditory goal associated with /u/ by altering their tongue position. In the present study, the importance of sensory feedback was further investigated by combining lip-tube and auditory feedback perturbations. Fifteen adult francophone speakers produced 5 blocks of ten /u/ tokens under various conditions (following and preceding a baseline condition). First, tokens were produced when the speakers had a lip-tube in place (predicted to increase F1 and F2) as well as a real-time auditory feedback perturbation (designed to cancel the acoustic effects of the lip-tube, that is, to decrease F2 and F1). Next, only the lip-tube perturbation was applied, with and without white noise. Formant values for each condition were extracted at the vowel midpoint. Although most speakers produced a larger compensatory response with the lip-tube alone condition than with the combined lip-tube and auditory perturbation condition, all speakers produced altered formants in the latter condition. This suggests that speakers try to reduce auditory and somatosensory feedback

errors during speech production, with a speaker-specific weighting of each sensory modality.

5aSC16. Effects of focus and tone on vowel space in Chongming Chinese. Yike Yang, Bei Li, and Si Chen (Dept. of Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., AG511, Kowloon, Hong Kong, yike.yang@connect.polyu.hk)

The formant frequencies of vowels are prone to change according to contexts, but the effects of focus on vowel space are rarely studied. This study attempts to investigate how focus and tone may influence the vowel space in Chongming Chinese with a production experiment. Chongming Chinese is a Chinese dialect with a complex tonal system. Twelve syllables with the vowel /æ/ were chosen and the syllables spread over eight lexical tones (two level tones, four contour tones, and two checked tones). The target syllables were embedded in carrier sentences with different preceding and following syllables. Four focus conditions were manipulated: broad focus, initial focus, medial focus, and final focus. During the experiment, twelve participants were elicited with precursor questions to produce sentences with different focus locations. The frequencies of the first two formants (F1 and F2) were extracted over the middle 50 ms of the vowel interval for each target syllable. Formant tracks were calculated with the Burg algorithm in Praat. F1 and F2 values under each focus condition and tone will be compared. It is hypothesized that F1 and F2 will be increased under focus because focus might make the vowels more open and front.

5aSC17. Tahrir patterns in a professional Iranian classical singer. Mahdi Tahamtan and Ronald Scherer (Commun. Sci. and Disord., Bowling Green State Univ., Bowling Green, OH 43403, mahdit@bgsu.edu)

"Iranian classical singing" ("avaz" in Persian) contains the ornament called a tahrir. It typically is a sequence of tekyeh fundamental frequency gestures that quickly increase and then decrease. The primary aim was to determine and describe the tahrir patterns produced by a professional avaz singer. An unaccompanied recording was made of a professional avaz singer singing the popular song "Morghe Sahar" ("Dawn Bird"). Consistent patterns of tahrir productions were determined after repeated listening. Four primary tahrir patterns were identified, based on (1) the number of tekyehs within the tahrir, (2) fundamental frequency extent from the baseline to the tekyeh peaks, and (3) inter-tekyeh interval durations. Pattern 1 consists of one tekyeh and is called a "zinat." Pattern 2 is a multiple tekyeh gesture with relatively long inter-tekyeh intervals. Pattern 3 is a multiple tekyeh gesture with relatively short inter-tekyeh intervals. Pattern 4 is a vibrato-like pattern with multiple tekyehs with short inter-tekyeh intervals like Pattern 3 but with relatively short frequency extents. The results are pertinent to voice scientists interested in the mechanics of phonatory production, to singing teachers and artists interested in pedagogy based on acoustic and perceptual information, and to ethnomusicologists interested in cultural musical performance and production.

5aSC18. Nasal consonants block /o/ fronting in white and Lumbee speakers from Robeson County, North Carolina. Marie Bissell (Linguist., North Carolina State Univ., 2211 Hillsborough St., Tompkins Hall, Box 8105, Raleigh, NC 27607, marie.bissell@gmail.com)

This study examines /o/ fronting in white and Lumbee speakers in a triethnic contact situation in Robeson County, North Carolina. Previously, Thomas (1989) observed that following nasal consonants may be blocking /o/ fronting in Wilmington, North Carolina. The present study examines this hypothesis in more depth in nearby Robeson County. Nine speakers were selected from the Sociolinguistic Archive and Analysis Project (Kendall, 2010). Nucleus and glide measurements were taken for each speaker's /o/ tokens, pre-lateral /o/ tokens, and pre-nasal /o/ tokens. Values were then Lobanov-normalized. Acoustic analysis of individuals speakers' vowel spaces suggests that when /o/ fronts, pre-nasal /o/ remains back along with pre-lateral /o/. While white speakers' pre-nasal /o/ patterns closely with their pre-lateral /o/, Lumbee speakers' pre-nasal /o/ tends to glide in a different direction in what could be considered a three-way allophonic split in the /o/ vowel class. The results of this study indicate that both following lateral consonants and following nasal consonants block /o/ fronting, though the phonological mechanism underlying pre-nasal /o/ backing is in need of further investigation.

5aSC19. Coarticulatory resistance in Assamese stops. Indranil Dutta (Dept. of Computational Linguist., The EFL Univ., Hyderabad, Telangana, India), Pamir Gogoi, and Ratree Wayland (Dept. of Linguist., Univ. of Florida, Gainesville, FL 32611, pgogoi@ufl.edu)

The nature and extent of coarticulation and coarticulatory resistance of speech segments have been found to be dependent on the size, shape, and density of contrasts, in addition to neighborhood densities of segment sequences. While high density of contrasts leads to high coarticulatory resistance, low density of contrasts leads to low coarticulatory resistance. Assamese, an Indo-Aryan (IA) language, unlike the other IA languages which exhibit a two-way coronal place contrast, contains only an alveolar stop. Due to the relative low density of the alveolar, we hypothesize that the Assamese alveolar stops will exhibit low coarticulatory resistance compared to the velar counterpart. We use locus equations to analyze the coarticulatory effects. First order locus equations were used in a $[_{aCV_}]$ context, where C is a labial, alveolar, or velar stop, while the following V is one of the corner vowels /a,i,u/. Unlike the hypothesis, the least amount of coarticulatory resistance is shown by the velars and the alveolars show greater resistance compared to the velars and the labials. This poses an interesting question because it has been stated that mostly the slopes are seen to be order of labial, then velar and then alveolar. The possible explanations are discussed.

5aSC20. Analyzing rhythmic convergence between two Swiss German dialects. Elisa Pellegrino (URPP Lang. and Space, Freiestrasse 16, Zurich 8032, Switzerland, elisa.pellegrino@uzh.ch) and Volker Dellwo (Dept of Computational Linguist., Zurich, Switzerland)

With respect to existing evidence of rhythmic adjustments in response to the interlocutor's idyosincratic characteristics, in the present study, we test whether interlocutors are likely to mutually adapt their rhythmic characteristics over the course of a conversation or after increased exposure to a dialogue partner. To study rhythmic accommodation, we used a corpus of read speech recorded by 18 speakers of two Swiss German dialects—Grison and Zurich German—before and after performing diapix tasks. The two dialects present crucial differences in the durational characteristics of intervocalic sonorants, vowel in open syllable and vowel in final position, thus creating different rhythm in these two dialects. To determine whether Grison and Zurich German speakers produce the rhythmic contrasts more similarly after the diapix tasks, we measured the Euclidean distance within a pair (*ddpair*) and within an individual (*ddspeak*) in three ratio measures devised to capture the durational differences between the two dialects. Preliminary results on *ddpair* have shown that certain durational contrasts are produced more similarly by certain pairs but more differently by other ones. Ratio measure related to the timing properties of vowels in final position does not change drastically before and after the interaction.

5aSC21. Effect of stress, harmonic status, and lexical status on Hungarian vowel-to-vowel coarticulation. Jenna Conklin and Olga Dmitrieva (Linguist., Purdue Univ., 610 Purdue Mall, West Lafayette, IN 47907, jconkli@purdue.edu)

The present study examines vowel-to-vowel coarticulation in Hungarian with particular emphasis on the effects of word stress and lexical status (real words versus nonce words) on the direction and degree of vowel coarticulation. Previous investigations indicate that lexical stress exerts a powerful influence on vowel coarticulation. Specifically, stressed vowels are thought to resist coarticulation (see, e.g., Beddor *et al*., 2002; Majors, 2006). It is not clear, however, whether this effect is universal across languages, nor if it can be suppressed by the presence of more influential factors. In addition, it is not known whether vowel-to-vowel coarticulation always affects real and nonce words in a similar fashion, though some previous research indicates that speakers differ in their coarticulatory handling of nonce words (Scarborough, 2012). Finally, previous research suggests that the presence of vowel harmony in a language may place restrictions on the direction of vowel-to-vowel coarticulation (Beddor and Yavuz, 1995). Hungarian has a vowel harmony system that includes several neutral vowels, which do not participate in harmony. This study reports the results of an acoustic analysis

of coarticulation in stressed and unstressed vowels (both neutral and harmonizing) in real and nonce words produced by native speakers of Hungarian.

5aSC22. Variation in voice quality within speakers. Yoonjeong Lee and Jody E. Kreiman (Head and Neck Surgery, Univ. of California, Los Angeles, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeonglee@ucla.edu)

Little is known about the nature or extent of everyday variability in voice quality within a speaker or how this differs across speakers. Using a suite of measures that map between acoustics and perception of voice quality, this study elucidates which acoustic variables within speakers' individual voice spaces best characterize speakers. Based on studies of faces and cognitive models of speaker recognition, we hypothesized that a few measures would be important across speakers, but that much intra-speaker variability would be idiosyncratic. By using principal component analysis, we tested this hypothesis against a set of multiple sentence productions from 100 native speakers of English (fifty females and fifty males), recorded over three days. Acoustic variables measured every 5 ms on vowels and approximants corresponded to F0, vowel quality, spectral noise, source spectral shape, and variability. Across speakers the balance between higher harmonic amplitudes and inharmonic energy in the voice accounted for the most variance (females = 20%/males = 22%). Vowel quality and its variability accounted for an additional 12%/12% of variance. Remaining variance appeared largely idiosyncratic, suggesting that the speaker-specific voice space is different for different people. Notably, F0 did not emerge from these analyses. Implications for voice recognition are discussed. [Work supported by NIH/NSF.]

5aSC23. Vowel coarticulation in the Buckeye corpus. Sean A. Fulop (Linguist., California State Univ. Fresno, 5245 N Backer Ave., PB92, Fresno, CA 93740-8001, sfulop@csufresno.edu) and Hannah Scott (Elec. Eng. and Comput. Sci., Oregon State Univ., Corvallis, OR)

Theories of vowel perception, production, and acquisition have moved on from the early simple assumption of a single target for each vowel in auditory space, and yet modern variations continue to employ this notion in more sophisticated garb. The vowel prototype is lately conceived as the result of an auditory processing function incorporating information about both vowel dynamics and flanking sounds. Vowel sounds most often are flanked by consonants which are coarticulated with the vowel. It is commonly presumed that listeners apply an unknown auditory transformation to undo the coarticulatory effects when processing speech cognitively. This presumption requires coarticulation to be systematically predictable from neighboring consonants, which some studies have not found to be the case. The present study examines the vowel formants of 35 speakers from the Buckeye corpus of American English. It is found that consonant-vowel coarticulation effects are not sufficiently systematic across speakers to be predictable from the consonants. Moreover, since the coarticulation effects are so large and ubiquitous, there are no "ideal" vowels to be found in the data. There appears to be no way to transform such acoustic data by relying on the consonants that could yield a set of ideal vowel prototypes.

5aSC24. Voice actors imitating child speech: A study using 3D ultrasound. Colette Feehan and Steven M. Lulich (Linguist., Indiana Univ., Bloomington, 107 s Indiana Ave., Bloomington, IN 47404, cmfeehan@indiana.edu)

Voice actors are an interesting population for linguistic study because their profession requires them to perform complex vocal tract manipulations in order to portray specific social identities and convey socially indexed linguistic information. Previous investigations have looked at how voice actors manipulate laryngeal setting and voice quality to portray specific character types in animation (Teshigawara and Murano, 2004; Starr, 2015), but investigations of articulatory manipulations employed by voice actors are rare. This study uses three-dimensional ultrasound, formant, and F0 analyses to compare the strategies that one amateur and one professional voice actor use in order to imitate child speech. Preliminary analysis shows that the amateur actor relies on manipulation of articulatory setting by implementing hyoid bone raising, gesture fronting, and tongue grooving in order to sound like a child. The professional actor relies more on manipulation of laryngeal structures and prosody. Despite these differences in approach, the two actors still achieve similar child-like percepts. This study will describe the differences in strategy implemented by each actor as well as the within-subject variation across each actor's adult and imitated child voices.

5aSC25. Tracking sources of formant frequency variation in data from a time-controlled speech production task. Stefon M. Flego (Linguist., Indiana Univ., Bloomington, 720 S College Mall Rd. Apt. N5, Bloomington, IN 47401, sflego@indiana.edu)

Vowel formant structure is particularly sensitive to influence from neighboring segments in connected speech. Statistical models can successfully account for acoustic variability from multiple simultaneous sources of variance, such as anticipatory effects of following consonants and vowels (see, for example, Cole *et al*., 2010). Statistical approaches have also been developed to predict context segments from static measures of spectral variation like mean or midpoint formant frequencies. This study examines shifts in the relative strength of these sources of variance over the course of the target vowel, and compares previous approaches to predicting the identity of upcoming segments with a novel approach which is cumulatively informed by formant frequency variation as it unfolds temporally. Data are from 32 participants who produced target-context word sequences (permutations of …V_1C#hV_2…, where V_1 = [ɛ], C = [θ f], and V_2 = [i ɛ æ ɑ u], e.g., "deaf-heating") within a carrier sentence. Timing and prosody were strictly controlled by entraining the syllable rate to a metronome. Context word onset was always [h], which has no intrinsic supralaryngeal articulation and only begins foot-initial syllables in English. This ensured a consistent prosodic boundary between target and context words and prevented resyllabification of C across words.

5aSC26. The acoustics of feeling: Emotional prosody in the *StoryCorps* corpus. Rachel M. Olsen (Linguist, Univ. of Georgia, 142 Gilbert Hall, Athens, GA 30602-6205, rmm75992@uga.edu)

Effective communication depends not only on lexical content but also on how language is produced and perceived. Prosodic elements such as intensity, pitch accent (i.e., the pattern of low and high tones used in a stressed word), and intonation, have been suggested to aid in conveying emotional affect (e.g., happiness) in acted speech. For example, the word "yes" produced with a falling tone, high intensity, and large pitch range may indicate a high-arousal emotion like anger, whereas the same word produced with a flatter intonation, and lower intensity may indicate a low-arousal emotion like gloominess. Understanding the emotion with which words are produced facilitates appropriate communication because it tells the interlocutor how to best respond. Although this is intuitively true, the role of prosody in conveying emotional meaning is understudied, particularly in natural speech. This project thus utilizes *StoryCorps,* an extensive corpus of naturalistic interviews (publicly available at www.storycorps.org), to acoustically analyze pitch-accent usage in speech conveying different emotional affects. Portions of the *StoryCorps* interviews that convey overt emotional effect are selected and transcribed, and f0 trajectory, pitch range, duration, and intensity are tracked across stressed vowels to explore whether there are distinctive pitch-accent patterns used to convey different emotions.

5aSC27. Phonetic convergence and divergence in the American Midwest. Cynthia G. Clopper and Ellen Dossey (Ohio State Univ., 1712 Neil Ave., Oxley Hall 100, Columbus, OH 43210, clopper.1@osu.edu)

The magnitude of phonetic convergence in word shadowing is affected by phonetic factors, including the shadowers' phonetic distance from the model talker, and social factors, including the perceived prestige of the model talker's variety. The current study explored the effects of these factors on phonetic convergence to the Northern dialect of American English by shadowers from the Midwestern United States. The shadowers read a set of target words to provide their baseline productions and then repeated the same target words after a native speaker of the Northern dialect. The target words contained stressed vowels that are phonetically shifted in the Northern dialect relative to other American English varieties. Phonetic convergence in word duration and vowel formant frequencies was observed for some vowel categories but not others. This phonetic selectivity was not driven by the acoustic distance between the shadowers' baseline productions

and the model talker's productions. Phonetic convergence in word duration and vowel formant frequencies also varied depending on whether or not the shadowers were told where the model talker was from, leading to divergence for some dialect-specific features in the shadowing task. These findings provide additional evidence for the roles of phonetic and social factors in phonetic convergence.

5aSC28. Sociolinguistic differences in production of pre-nasal /o/: Evidence from Southeastern Ohio. Peter A. Andrews (English, North Carolina State Univ., Tompkins Hall, Raleigh, NC 27606, paandrew@ncsu.edu)

This study investigates the production of /o/ among 14 young adults (ages 18–24) in Athens, Ohio. This vowel is phonetically atypical because it is subject to lexical and phonological conditioning among non-native Athenians living in Athens. To better understand the phonological effects of a following nasal on /o/, the speech of 14 native southeast Ohioans was analyzed acoustically. The data come from sociolinguistic interviews selected from the Southeast Ohio Language Project corpus (Lee *et al*., 2018). The first two formants of each token of /o/ were extracted (n = 1468), normalized (Lobanov, 1971), and plotted. A mixed effects model was applied to the normalized F2 data to uncover interactions between location (Athens versus non-Athens) and vowel class. Results of the analysis suggest that non-Athenians have a significantly higher F2 value for /o/ than Athenians, and further that non-Athenians distinguish between two /oN/ allophones depending on the lexical item, a distinction that Athenians seem to lack. Furthermore, this fronting effect is more pronounced in words that have a preceding post-alveolar or labiovelar consonant, both of which are likely to result in a higher F2 of the vowel. These findings are an important contribution toward a more thorough understanding of pre-nasal phonological conditioning.

5aSC29. Acoustic, non-invasive measurement of velopharyngeal aperture using a high frequency tone. Kevin B. McGowan (Linguist., Univ. of Kentucky, 1415 Patterson Office Tower, Lexington, KY 40506, kbmcgowan@uky.edu), Michael T. Johnson (Elec. and Comput. Eng., Univ. of Kentucky, Lexington, KY), Aleah D. Combs (Linguist., Univ. of Kentucky, Lexington, KY), and Mohammad Soleymanpour (Elec. and Comput. Eng., Univ. of Kentucky, Lexington, KY)

Many researchers need access to the real time articulatory state of the velopharyngeal port to investigate the timing and extent of nasal gestures alongside the acoustic consequences of those gestures. While numerous methods exist for the investigation of nasalization (e.g., airflow, velotrace, and nasometer), these methods tend to be invasive, expensive, or to muffle the acoustic speech signal in pursuit of nasal articulatory data. We describe an inexpensive method and procedure for the investigation of nasal gestures using low frequency ultrasound. This system injects a 20 kHz tracer tone into a nostril using inexpensive components and/or three-dimensional printed parts. This signal can then be collected along with the speech signal using microphones typical in speech research. We will compare the results of this system to those of a state of the art airflow collection system and compare cross-participant reliability of both methods.

5aSC30. A non-primary cue in spontaneous imitation of English voiceless stops. Harim Kwon and Yuting Guo (English, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, hkwon20@gmu.edu)

When speaker-listeners are exposed to an exaggerated primary cue, they spontaneously imitate the enhanced cue; e.g., English speakers imitate extended VOTs of voiceless stops (e.g., Nielsen, 2011). Kwon (2015) claims that speaker-listeners enhance the primary cue for the relevant contrast when exposed to an enhanced non-primary cue, showing that Seoul Korean speakers imitated aspirated stops with an enhanced non-primary cue (stop VOT) by exaggerating the primary cue for phonological aspiration in the language (post-stop f0). However, VOT in Seoul Korean, despite being a non-primary cue, may enjoy a special status, since it is required to maintain the full three-way laryngeal contrast in the language. This study examines how the non-primary cue enhancement is imitated when the cue is not necessary for the relevant contrast. Native speakers of American English heard and spontaneously imitated English /t/ with either extended VOT (the primary cue for English voicing contrast) or raised post-/t/ f0 (a non-primary cue). Participants' own productions of English /t/ and /d/ before and after the exposure were compared. Preliminary findings suggest that the participants produced /t/ with longer VOT and higher post-stop f0 after hearing the stimuli with either manipulation but to a lesser degree after the f0-raised stimuli than the VOT-extended stimuli. The role of a non-primary cue in spontaneous imitation will be discussed.

5aSC31. Respiratory and electroglottographic measures of normal and loud speech across vowels. Laura L. Koenig (Haskins Labs. and Adelphi Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu) and Susanne Fuchs (Leibniz-Ctr. General Linguist., Berlin, Germany)

Louder speech is primarily associated with greater respiratory system driving pressures, but several studies have indicated that speakers may also adjust their laryngeal settings in louder speech. In combination, these observations suggest the possibility of a tradeoff between respiratory and laryngeal mechanisms for increasing loudness, but little past work has obtained both respiratory and laryngeal measures in multiple speakers to assess the range of individual strategies. In this study, we assess measures of voice quality obtained from electroglottographic [EGG] signals in multiple female speakers. Respiratory kinematics, obtained by inductance plethysmography, were recorded simultaneously. Louder speech was obtained naturalistically, by changing speaker-interlocutor distance. Previous analyses of the respiratory data indicate that speakers differ considerably in how much they vary their respiratory kinematics when producing loud speech. We hypothesize that speakers who show less variation in their respiratory behavior will show more extreme changes in the EGG signals between normal and loud speech. We will also evaluate whether EGG differences between normal and loud speech vary as a function of vowel quality, in light of past work showing systematic differences in voice quality measures across vowels.

5aSC32. Applying refined automatic formant measurement to determination of the orientations of vowel distributions. Jeff Mielke, Erik R. Thomas (English, North Carolina State Univ., Box 8105, Raleigh, NC 27695, erthomas@ncsu.edu), Josef Fruehwald (Linguist. and English Lang., Univ. of Edinburgh, Edinburgh, United Kingdom), Jane Stuart-Smith (English Lang. and Linguist., Univ. of Glasgow, Glasgow, Scotland, United Kingdom), Morgan Sonderegger (Lingusitics, McGill Univ., Montreal, QC, Canada), Robin M. Dodsworth (English, North Carolina State Univ., Raleigh, NC), and Michael E McAuliffe (Lingusitics, McGill Univ., Montreal, QC, Canada)

In order to remedy recurrent problems with false formant readings obtained with an automatic formant measurement routine, prototype-based automatic measurements were compared with manual formant measurements of the same uttered vowels. Two refinements that avoided false formants, one involving the option to skip measured formant racks and the other involving an expectation that successive formants would show successively lower amplitudes, were developed. This method was then applied to seven corpora representing diverse English dialects, with satisfactory results. The measurements of each vowel thus obtained were then subjected to principle component analysis to determine the orientation of the tokens in F_1/F_2 space. Most vowels exhibited distributions that apparently reflect degree of jaw opening. However, /u/ in North American varieties (with pre-/l/ and post-/j/ tokens excluded) showed mostly horizontal orientations, even when post-coronal and non-post-coronal tokens were considered separately. This pattern contrasted sharply with the vertical orientations of mid back vowels. /u/ was also the only vowel whose orientations coincided consistently with ongoing changes in the communities. We hypothesize that the factors responsible for the horizontal orientations of /u/ also lie behind its cross-linguistic tendency to shift frontward.

5aSC33. A comparison of prosodic patterns in English bi-morphemic lexical and function words. Irina A. Shport and Gregory Johnson (English, Louisiana State Univ., 260-G Allen Hall, Baton Rouge, LA 70803, ishport@lsu.edu)

In English, monosyllabic function words (*them*, *in*) are prosodic clitics, but the prosodic status of multisyllabic function words (*whenever, everybody*) is not well understood. We compared fundamental frequency (F0) patterns in function and lexical compounds to determine whether the words

formation process, compounding, is associated with a certain prosodic template. F0 was examined because of vowel-intrinsic differences in duration and intensity, the other two correlates of prominence. Four speakers of Appalachian English from Kentucky read sentences with words of three types in the sentence-medial position: two relative pronouns (*wh**at**ever*, ***e**ver-wh**at***), two quantifiers (***e**very**o**ne*, ***e**very**o**ne*), and six lexical words (e.g., *w**all**fl**o**wer*, ***e**vergr**ee**n*). F0 ratios between stressed vowels in two morphemes of each word were calculated. The results showed that the ratios were higher in lexical compounds than in quantifiers and in free relative pronouns, which were not different from each other. These results suggest a pattern of the first-morpheme (left-constituent) prominence in lexical words, similar to previous reports on adjective + noun compounds, and a pattern of the second-morpheme (right-constituent) prominence in function words, similar to morphologically complex but non-compound words. The proposed right-prominence prosodic template for multi-morphemic function words is discussed in relation to phrase-level accents and boundary tones.

5aSC34. Where the skies are not Cl/aʊ/dy all day: A pilot study of /aʊ/ production of speakers in rural Kansas. Matthew J. Champagne (English, North Carolina State Univ., Rm. 204 Thompkins Hall, 2211 Hillsborough St., Raleigh, NC 27603, mjchampa@ncsu.edu)

This study examines cross-generational /aʊ/ production in rural, female speakers from Dickinson County, Kansas. Acoustic analysis of nuclei and glides were BARK normalized for cross-speaker, cross-generational comparison. This study finds older and younger speakers show similar nucleus advancement and glide direction.

5aSC35. Methods for noise reduction in a legacy speech corpus. Lisa Lipani, Yuanming Shi, Joshua McNeill, and Margaret E. Renwick (Univ. of Georgia, 142 Gilbert Hall, Athens, GA 30602, llipani@uga.edu)

The Digital Archive of Southern Speech is an audio corpus featuring interviews conducted from 1968 to 1983, with speech from 30 female and 34 male southern speakers, totaling 372 h of audio data. However, automated analysis of this data has been made difficult by background noise in this legacy corpus, originally recorded on reel-to-reel tape and later digitized to .wav format. In this paper, we compare the effect of noise reduction techniques on the acoustic signal and evaluate their effect on acoustic speech data. We use Praat to detect the quietest silence (measured using root-mean-square amplitude) in each sound file. The audio data contain silences that lack background noise (e.g., due to anonymization procedures), but these are excluded from selection. Each quietest silence is used to create a "noise profile," which is removed from the audio using a scripted noise removal procedure in both Audacity and SoX. The success of each procedure is assessed with three-way comparisons of the amplitude of silences in uncleaned and cleaned audio, and vowel plots made with formant values extracted from uncleaned and cleaned audio. It is predicted that successful noise reduction will reduce the number of outliers occurring in F1, F2 space.

5aSC36. An articulatory analysis of asymmetrically confusable consonants. Ian Calloway (Linguist., Univ. of Michigan, 400 Lorch Hall, 611 Tappan Ave., Ann Arbor, MI 48109, iccalloway92@gmail.com)

Acoustic similarity and articulatory similarity do not always co-occur; a narrow range of acoustic outcomes can sometimes correspond to a comparatively wide range of articulatory parameters (Stevens, 1989). The articulatory parameters associated with a narrow window of acoustic events occasionally straddle a category boundary—some consonants (e.g., /t/ and /k/ in front vowel contexts) show acoustic (and perceptual) similarity with one another, despite differences in active articulator or place of articulation (Plauché, 2001). This study utilizes a corpus of vocal tract MRI (Sorensen *et al*., 2017) to relate the spatial dynamics of productions of /p/, /t/, /k/, /f/, and /θ/ to their acoustic properties. This analysis offers insight into why articulatorily dissimilar productions can show increased spectral similarity and how phonetic context conditions this similarity. For each segment production, five video frames were extracted—one frame before constriction release, and four after release. From each frame, a 30-point cross-sectional area function of the vocal tract airway was generated. Vocal tract parameters associated with the spectral properties of frication, aspiration, and stop bursts were analyzed. In phonetic contexts favoring misidentification, confusable segment pairs show greater similarity in vocal tract regions anterior to the primary constriction.

5aSC37. *Kyoo*, what an odd-sounding word! Acoustic analyses of a Cajun interjection. Lauren V. Vidrine and Irina A. Shport (Louisiana State Univ., 5151 Highland Rd., Baton Rouge, LA 70808, lvidri5@lsu.edu)

The interjection *kyoo* is used in English and French spoken by Cajun heritage speakers in Louisiana to express surprise. Cajun and non-Cajun speakers alike comment that *kyoo* sounds "weird," as if it has some non-English characteristics. These perceptions may stem from Cajun French influence on Cajun English. Specifically, nasalized vowels and unaspirated stops are reported to occur in this dialect, more frequently in older than younger speakers. This case study investigated *kyoo* productions in two groups of Cajun English speakers to determine whether the word conformed to French-influenced sound patterns. Acoustic analyses showed that despite speaker auditory perceptions, the *kyoo* vowel was not nasalized as compared to other English words with oral and nasal codas. The spectral quality of this vowel was most similar to /ɔ/. Furthermore, voice onset time of the initial /k/ was not consistent with that of unaspirated stops. These results suggest the perception of anomalous sounds in this interjection may not be attributed to the acoustic characteristics indicative of French influence. Alternative explanations may include relatively long vowel duration due to speakers' enthusiasm to perform Cajunness, palatalization of the initial stop followed by the open-mid back vowel, or simply non-standard nature of *kyoo* stereotyped by speakers.

5aSC38. Definite determiner realization in the Philadelphia neighborhood corpus. Jennifer B. Arlin (Linguist., Univ. of Pennsylvania, 3401 Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104-6228, jarlin@sas.upenn.edu)

Systematic variation in the pronunciation of the definite determiner (orthographic *the*) is well-known; the general rule is that preconsonantal /ðə/ alternates with prevocalic /ði/. Nonetheless, widespread deviation from this rule has been noted, most recently by Meyerhoff (2018) (examining definite determiner pronunciation in various populations in New Zealand). This project studied *the* realization among a small sample of female speakers in the Philadelphia Neighborhood Corpus. All were L1 English speakers, born between 1950 and 1986 and interviewed between 1973 and 2010. The determiner was hand-coded throughout their interviews for pronunciation and for nature of the following segment (vowel- or consonant-initial, as uttered). The data were then analyzed to determine whether age or following segment influenced realization as /ðə/ or /ði/. Results indicated a high degree of adherence to the canonical rule, showing preconsonantal /ðə/ 97% of the time and prevocalic /ðə/ 18% of the time. While the rate of occurrence of prevocalic /ðə/ was generally quite low, it did vary significantly by speaker. The younger generation on average tended to use prevocalic /ðə/ slightly more often than the older generation; individual variations made it impossible to draw any conclusions about a meaningful connection between age and variation.

5aSC39. Reference speaker selection for kinematic-independent acoustic-to-articulatory-inversion. Narjes Bozorg and Michael T. Johnson (Elec. and Comput. Eng., Univ. of Kentucky, 1608 University Court, Unit C312, Lexington, KY 40503, narjes.bozorg@uky.edu)

This paper investigates the most effective reference speaker set for the Parallel Reference Speaker Weighting (PRSW) algorithm for kinematic-independent acoustic-to-articulatory inversion. To obtain the adaptation weights for estimating the articulatory model, different reference speaker accent types and quantities have been acquired. The reference speaker sets have been selected not only based on their performance in speaker-dependent kinematic-inversion but also based on the type of accent. The experiments have been conducted on parallel Acoustic-Articulatory data, the Marquette Electromagnetic Articulography corpus of Mandarin Accented English (EMA-MAE) consisting of 20 native English speakers and 19 native Mandarin speakers of English. A comparison is made between different types of target speakers and reference speakers with results indicating that the accuracy of the adapted model increases when we select balanced distributed accents of English and lower number of speakers.

5aSC40. Individual differences in the production of prosodic boundaries in American English. Jiseung Kim (Univ. of Michigan, 440 Lorch Hall, Ann Arbor, MI 48109, jiseungk@umich.edu)

This study investigates individual differences in the weighting of phonetic properties in the production of prosodic boundaries in American English. The motivation of the study is to inform understanding of individual speaker variation and its accommodation in the representation of prosodic structure. In an acoustic study, 32 speakers produced eight sentence pairs differing in type of boundary (Intonational Phrase (IP) boundary versus Word boundary). Pause duration, phrase-final lengthening (three syllables before the boundary), phrase-initial lengthening (one syllable after the boundary), and pitch reset were examined. The results of a series of statistical analyses showed substantial individual differences in (1) which segmental and suprasegmental properties speakers phonetically modulated to produce IP boundaries and (2) the scope and the degree of such modulations. In addition, the results suggest that there seems to be no apparent relationship between how speakers modulated boundary-adjacent syllable durations and whether and how they used two other acoustic correlates for IP boundary, namely, pause duration and pitch reset. The current models of prosodic structure need to accommodate the fact that individuals may vary significantly while systematically modulating the acoustic correlates relevant for encoding a prosodic contrast.

5aSC41. Semantic predictability and the use of creaky voice in female speakers. Stefania Marchitelli (Communicative Sci. and Disord., New York Univ., 6 Old Homestead Way, Albertson, NY 11507, sm6276@nyu.edu) and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

The purpose of this study was to investigate the prevalence of creaky voice in sentence-final position as a function of semantic predictability of the final word. Creaky voice is maligned in the media and often claimed to be less intelligible than normal phonation. A recent study finds some evidence to support this claim by examining intelligibility of single words. If speakers were concerned about intelligibility of their speech, they may be less likely to use it in sentences with little semantic support (e.g., "Mr. Black knew about the pad" compared to "Tear off some paper from the pad"). In the current study, 11 young female speakers without any vocal pathologies produced sentences from the Speech Perception in Noise (SPIN) test. Fourteen final words that appeared in both high and low semantic predictability sentences were selected for analysis. Three outcome measures were examined for these final words: presence versus absence of creaky voice, duration of creaky voice, and type of creaky voice. Contrary to our expectations, the results indicated that semantic support does not significantly predict the existence, amount, or type of creaky voice in young female speakers.

5aSC42. Transfer in speech motor learning: The role of voicing. Hung-Shao Cheng (Communicative Sci. and Disord., New York Univ., 2559 35th St. Apt. 2L, Astoria, NY, hscheng@nyu.edu) and Adam Buchwald (Communicative Sci. and Disord., New York Univ., New York City, NY)

Previous studies have demonstrated that American English speakers can improve their production of phonotactically illegal onset clusters (e.g., DBEEGO) after structured practice. However, the nature of what is learned remains incompletely understood. We use a transfer paradigm to address this question by examining performance on trained and untrained novel consonant sequences. In particular, we investigated whether the differences between voiced and voiceless stop-stop clusters (e.g., /gd/ vs. /kt/) influences transfer of learning, hypothesizing that the voiced clusters involve more complex motor control. Forty native speakers of American English practiced nonwords beginning with either voiced (/gd/, /db/, /gb/) or voiceless (/kt/, /kp/, /tp/) stop-stop onset clusters. All participants were tested on both types of clusters at baseline (prior to practice) and in two retention sessions (20 minutes (R1) and 2 days (R2) after practice). Blinded coders rated cluster accuracy based on presence of a vowel in the acoustics. Preliminary results (n = 10) indicate a trend of bi-directional transfer, with participants in both practice conditions exhibiting improved accuracy for both trained and untrained clusters.

5aSC43. Studying the rate of velar elevation across different vowel contexts in normal speech using high speed nasopharyngoscopy. Hedieh Hashemi Hosseinabad (Commun. Sci. and Disord., Eastern Washington Univ., 310 N Riverpoint Blvd., Box B, Spokane, WA 99202-1609, hhosseinabad@ewu.edu), Liran Oren (Univ. of Cincinnati, Cincinnati, OH), Ann W. Kummer (Cincinnati Childrens Hospital, Cincinnati, OH), and Winter Taite (Washington State Univ., Spokane, WA)

Velum has a range of positions across different speech segments, with the lowest velum position for nasal consonants, a high (closed) position for obstruent consonants, and a range of positions in the middle for sonorant consonants and vowels varying according to constriction degree. Vowel environment has a considerable influence on the rate of velum closure. Changes in the rates of velar positioning during production of vowels in non-nasal speech (H-words) was studied in seven typically speaking adults with general American dialect and normal oral-nasal resonance. The participants were scoped using a Phantom Miro 310 high-speed video camera (Image acquisition rate of 5000 fps) connected to a flexible scope. Measurements were taken simultaneously with capturing acoustic data. The data suggested that velum tends to have a higher elevation point for vowels with higher degree of constriction in the oral cavity like /i/. Further results will be discussed.

5aSC44. Relative influences of information structure and utterance-final position on the prosodic implementation of nuclear pitch accents. Eleanor Chodroff and Jennifer Cole (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, eleanor.chodroff@gmail.com)

The phonological and phonetic realization of a nuclear pitch accent has been claimed to reflect aspects of its information structure (IS). As the rightmost accented word in an intonational phrase, the nuclear pitch accent often co-occurs with utterance-final position, which in American English, is often cued by prosodic means such as creaky voice and domain-final lengthening. The present study investigated the relative influence of IS and utterance-final position on the prosodic implementation of words in nuclear position. The IS of an object noun phrase occurring in nuclear position was manipulated to be given, accessible, new, or contrastive relative to a parallel object noun in the preceding sentence. The critical object noun occurred in utterance-final position in experiment 1 and in non-final position and preceding a semantically vacuous syntactic phrase in experiment 2 (e.g., "that was there," "for it). Given that contrastive information significantly influenced the prosodic implementation of the object noun in both experiments with respective reduction and enhancement effects, but utterance-final position nevertheless regulated the particular phonetic instantiation. While IS modulated the degree of creakiness, duration, and intensity in final position, IS conditions in non-final nuclear position modulated the type of f0 contour and duration.

5aSC45. The relationship between duration and spectral position in white southern US speech. Jon Forrest (Linguist., Indiana Univ., 830 Ballantine Hall, Bloomington, IN 47405, jonforr@iu.edu)

Previous research on regional differences in vowel quality and duration found that white Southern speakers make less of a distinction between high and mid tense-lax vowel pairs in both quality and duration than speakers from other regions (Kendall *et al*., 2014). However, the effect of duration on vowel quality in Southern speech has not been explored to the same degree. This study examines duration, vowel quality (in F1-F2 space), and their relationship. Data come from 80 h of self-recorded and interview speech from 17 white Southern natives with varying degrees of the Southern Vowel Shift (SVS). Measurements for F1 and F2 were extracted at vowel nucleus (25% duration) for the /e/-/ɛ/ and /i/-/ɪ/ pairs both implicated in the SVS. Formants were Lobanov-normalized and both duration and vowel quality were examined with mixed-effects linear regression. Results show that most speakers maintain a durational distinction between tense-lax pairs, but the effect of duration on vowel quality varies greatly from speaker to speaker. Those speakers who exhibit higher degrees of SVS shift show individual slopes for duration in the opposite direction than less-Southern counterparts. First, these results demonstrate that controlling for individual differences in duration is important for properly analyzing Southern speech corpora.

Second, they suggest that duration, vowel quality, and their relationship all play a role in regional distinction.

5aSC46. Acoustic characteristics of aspirated fricatives in a Tibeto-Burman language. Maureen Hoffmann (Dept. of Linguist., Univ. of Arizona, 402 S Reisner Dr., Bloomington, Indiana 47401, mhoffm@email.arizona.edu)

Aspirated fricatives are typologically rare, though they are relatively more common in Asia (Jacques, 2011). Of the 2155 languages in the PHOIBLE Online database of phonological segments, only 7 are reported to have distinctive fricative aspiration (Moran *et al*., 2014). This paper examines the acoustic properties of aspirated fricatives in Matu, an underdocumented Kuki-Chin (Tibeto-Burman) language spoken in Chin State, Burma (Myanmar). Matu contrasts unaspirated alveolar fricative /s/ with aspirated alveolar fricative /s^h/. Aspirated alveolar fricatives are also found in other Chin languages and have been reconstructed for Proto-Kuki-Chin (VanBik, 2009, p. 186). The data were recorded with Matu-speaking refugees living in Indiana, U.S. and are analyzed using durational and spectral measures. This research expands literature on an uncommon—and reportedly diachronically unstable (Jacques, 2011)—phonemic contrast while increasing documentation of an understudied language. Jacques, G., "A panchronic study of aspirated fricatives, with new evidence from Pumi," *Lingua*121, 1518–1538 (2011). and Wright, R.(eds). 2014. *PHOIBLE Online*, edited by S. Moran, D. McCloy, and R. Wright (Max Planck Institute for Evolutionary Anthropology, Lepzig, 2014). VanBik, K., "Proto-Kuki-Chin: A reconstructed ancestor of the Kuki-Chin languages," *STEDT Monograph Series #8* (University of California, Berkeley, CA, 2009).

5aSC47. A comparison of four vowel overlap measures. Matthew C. Kelley and Benjamin V. Tucker (Linguist., Univ. of Alberta, 4-23 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, mckelley@ualberta.ca)

Measures of vowel overlap to explore the acoustic similarity between proposed and existing vowel categories. They typically compare F1 and F2, and sometimes duration. In the present study, we investigate four methods of quantifying vowel overlap: the spectral overlap assessment metric (Wassink, 2006), the *a posteriori* probability-based metric (Morrison, 2008), the vowel overlap assessment with convex hulls method (Haynes and Taylor, 2014), and the Pillai score as used by Hay *et al*. (2006). Based on the data for /i/ and /ɪ/ in the dataset of Hillenbrand *et al*. (1995), we used Monte Carlo style simulations and repeated subsampling to assess each method. We examined both the two-dimensional (F1 and F2) and three-dimensional (F1, F2, and duration) versions of the methods. We took the methods' outputs as accurate if they produced values close to expected target values for each type of simulation, and we took the results as precise if there was little spread among the output values. The results suggest that the *a posteriori* probability-based metric is the most generally applicable, while the Pillai score should be used in scenarios where sensitivity to complete overlap is needed or where data cannot be said to be normally distributed.

5aSC48. Acoustic differences between normally phonated and whispered speech. Nichole Houle and Susannah V. Levi (New York Univ., 665 Broadway, New York, NY 10012, nh1473@nyu.edu)

Whispered speech is different from normally phonated speech in more than just lack of vocal fold vibration. Previous research suggests that speakers also modify their speaking rate and formant frequencies. The extent of these differences may depend on speaker gender, but this has not been studied in American English. We examined differences in duration and acoustic vowel space between normally phonated and whispered speech. We recorded 17 ciswomen, 11 cismen, and 10 transwomen producing hVd words in both normally phonated and whispered speech. Formant frequencies for /æ/, /ɪ/, /ɑ/, and /u/ were bark-transformed and vowel space area was calculated. The results indicated differences in the formants across the two speaking conditions, both in the absolute shift of formant values and in the size of the vowel space. In addition, differences in speaking rate (measured as the duration of the /hV/ portion) also showed differences between the two speaking conditions across the three speaker groups.

5aSC49. Temporal structures of English and Mandarin running speech. Can Xu, Mingshuang Li, Chang Liu (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX 78712, canxu@utexas.edu), and Sha Tao (National Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Beijing, China)

A number of studies have found that temporal envelope cues are significant for both English and Mandarin speech perception. However, native English and Mandarin speakers appear to differ in abilities to use these cues in speech recognition. The current study thus aims to investigate whether the temporal envelopes of English and Chinese running speech have any significant difference. Conversational speech of 16 males and 16 females from the United States and mainland China was recorded. The temporal envelope and long-term spectrum were analyzed and compared across the two groups of speakers. Results showed that in the temporal modulation domain, English speech had a peak at around 3 Hz, while for Chinese speech, the peak was observed relatively steadily from 3 to 5 Hz. Moreover, English speech possibly contained significantly longer and deeper temporal troughs to make use of than Chinese. [Works supported by the China National Natural Science Foundation (31628009).]

Session 5aUW

Underwater Acoustics: Underwater Signal Processing and Applications

David F. Van Komen, Chair
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Contributed Papers

9:00

5aUW1. Development of an in-air circular synthetic aperture sonar system as an educational tool. Thomas E. Blanford, John D. McKay, Daniel C. Brown, and Joonho D. Park (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, teb217@psu.edu)

Synthetic aperture sonar (SAS) is an underwater acoustic imaging technique that is used to generate high quality imagery through the coherent combination of signals measured on a moving platform. The practical challenges of underwater experimentation and the lack of publicly available data sets make it difficult to teach SAS processing to students using experimental data. These challenges are compounded by the complicated signal processing that is required to compensate for the unwanted platform motion that is common in field data. This presentation will detail the development of an in-air circular SAS system using commonly available laboratory equipment. By operating in air and using a circular geometry many of the challenges that students would otherwise encounter in SAS processing are either well controlled or completely eliminated. Imagery collected on the system will be displayed to demonstrate how meaningful data can be generated using a simple set of equipment. A number of applications of the system as a tool to teach array signal processing, pulse compression, and imaging concepts will be discussed as well.

9:15

5aUW2. Evaluation of underwater acoustic images obtained by the early stage monotone pulse-echo Synthetic Aperture Sonar. Kyungmin Baik (Ctr. for Medical Convergence Metrology, Korea Res. Inst. of Standards and Sci., Daejeon 34113, South Korea, kbaik@kriss.re.kr), Seung-Soo Park (SonarTech, Busan, South Korea), and Joong Eup Kye (Ministry of Trade, Industry and Energy, Seoul, South Korea)

Synthetic Aperture Sonar (SAS) is a side-looking underwater acoustic imaging sonar that can obtain higher resolution images than those by the conventional Side Scan Sonar (SSS). Due to its high resolution images, it can be applied to various fields of studies from military to civilian purposes. Republic of Korea recently has launched a research project developing the prototypes of SAS being operated in a few hundreds kHz of acoustic signals. This project adopted the towfish as the SAS platform that are to be towed under a few tens of water depth, which has a price advantage over Autonomous Underwater Vehicle (AUV), but gives unstable motion. It also adopted the monotone pulse-echo mode as transceiver method of the prototypes that gives relatively simpler implementation of the data acquisition module although it gives inferior SNR to the chirp mode. In the current study, the underwater acoustic images of the protype SAS taken both in the fresh water and offshore of Korea are to be shown. Current study also shows the evaluation method of image resolutions using test targets simulating point scatterers. Resulting images are to be compared with the images by the conventional SSS through the evaluation of the resolution. As is well known, although SAS generated clearer underwater images than SSS images, phase uncertainty of the array is critical factor determining the image resolution of SAS that is heavily degradated by the unwanted motion of the SAS platform.

9:30

5aUW3. Analysis of a ray-based blind deconvolution algorithm on ships of opportunity in the Santa Barbara Channel. Nicholas C. Durofchalk (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Rm. 131, Atlanta, GA 30332, ncd001@lvc.edu), Xuedong Zhang (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The ray-based blind deconvolution (RBD) technique for ocean waveguides estimates both the Channel Impulse Response (CIR) and the unknown waveform radiated by some source of opportunity measured on an array of hydrophones using only knowledge of the array geometry and the local sound speed. This method has been shown to be well suited for estimating CIRs in environments with water column depths of ~100 m and the estimated CIRs have been successfully used in conjunction with other techniques, such as matched field processing or waveguide invariant based approaches, for source localization, bottom parameter inversion, and other related studies. With the recent 2016 at-sea passive noise experiment in the Santa Barbara Channel (SBC), plenty of processing remains to be done; however, the RBD's performance and robustness in such an environment (~580 m depth, downward refracting profile) is not well documented. Here, we seek to investigate the feasibility and effectiveness of the RBD method in estimating CIRs from ships of opportunity in the SBC in terms of the assumptions of the technique, application to data, and numerical simulation.

9:45

5aUW4. A FNN for source range and ocean environment classification using time-domain features. David F. Van Komen (Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, david.vankomen@gmail.com), David P. Knobles (KSA, LLC, Austin, TX), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Mohsen Badiey (Univ. of Delaware, Newark, DE)

Acoustic source ranging in an uncertain ocean environment is a complicated problem, though classification and regression-based machine learning algorithms show promise. A feedforward neural network (FNN) has been trained to do either classification or regression on both the source-receiver range and environment type using extracted time-domain features. Time waveforms are generated to simulate signals received at different ranges in three different environments with a sandy, muddy, or mixed sediment bottom. Four features are extracted from these waveforms: peak level, integrated level, signal length, and later decay time. These four features are used to train FNN for both classification and regression of range and environment type, and the results are compared to a network trained on the time waveforms. For small amounts of training data, the extracted features provide a higher accuracy than the full waveform. Thus, physics-based feature selection via preprocessing can lead to fairly accurate results when using a FNN with small datasets. These results lay a foundation for comparisons to the more computationally expensive convolutional neural networks. [Work supported by the Office of Naval Research.]

10:00

5aUW5. A waveform-based convolutional neural net for source range and ocean environment classification. David F. Van Komen, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, david.vankomen@gmail.com), David P. Knobles (KSA, LLC, Austin, TX), and Mohsen Badiey (Univ. of Delaware, Newark, DE)

Neural networks learn features that are useful for classification directly from a source, such as a recorded signal, which removes the need for feature extraction or domain transformations necessary in other machine learning algorithms. To take advantage of these benefits and have a finer temporal resolution than a spectrogram, we built a one-dimensional convolutional neural network to classify source range and ocean environment from a received signal. The neural network was trained on simulated signals generated in different environments (sandy, muddy, or mixed-layer sediment layers) for several range classes. We found significant potential in a neural network of this type, given a large amount of varied training samples for the network to learn important features to make range and environment predictions. This type of network provides an alternative for frequency-domain learning and is potentially useful for impulsive sources. Benefits of using a time-domain envelope are also explored. Success in the time domain also reduces the computational requirements of conversion to frequency domain and increases the temporal resolution, which might be beneficial for real-time applications. [Work supported by the Office of Naval Research.]

10:15

5aUW6. Investigating parameter importance for different ocean environments using Fisher Information. Makenzie B. Allen, David F. Van Komen, Tracianne B. Neilsen, Mark K. Transtrum (Phys., Brigham Young Univ., N283 ESC, Provo, UT 84602, allenmakenzie1427@gmail.com), and David P. Knobles (KSA, LLC, Austin, TX)

In machine and deep learning, we often seek a simple and effective model. Overly complex models may be difficult to train and make inaccurate predictions. One way to find an effective model is to consider the relative impact of parameters on predictions as seen in the Fisher Information. The applicability of the Fisher Information is shown using numerically modeled transmission loss as a function of frequency for different range-independent ocean environments. The sensitivities to source-receiver range and depth and environmental parameters are quantified by calculating the Fisher Information. First, the Jacobian matrix of partial derivatives of transmission loss with respect to each of the model's input parameters is obtained, then the Jacobian matrix is used to calculate the Fisher Information matrix. An eigenvalue decomposition of the Fisher Information matrix shows that this system is "sloppy," because it exhibits an exponential hierarchy of parameter importance. In many cases, only a small number of parameters are relevant for explaining the model output but the impact of individual geoacoustic parameter varies with both environment and frequency. Our results have implications for learning algorithms and data collection methods while elucidating the relevant physics for different conditions. [Work supported by the Office of Naval Research.]

10:30–10:45 Break

10:45

5aUW7. Replicating a below-band linear field in a refracting sound channel via a caustic-corrected autoproduct. David J. Geroski (Randall Lab., Appl. Phys., Univ. of Michigan–Ann Arbor, 450 Church St., Ann Arbor, MI 48109, geroskdj@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Frequency difference source localization methods have achieved some success in uncertain underwater environments by taking advantage of the frequency difference autoproduct—a quadratic product of two complex field amplitudes at different frequencies but from the same location and recorded signal. The phase of the frequency difference autoproduct is less sensitive to unknown propagation complexities than that of the in-band field from which it is derived. The noted localization success arises in some technologically relevant scenarios when the frequency difference autoproduct at least partially mimics a genuine acoustic field at the difference of the two frequencies. However, the nonlinearity inherent in the autoproduct can lead to salient and even counter-intuitive differences between it and a genuine below-band field in the same environment. This presentation explores methods for correcting the phase differences that arise between autoproducts and genuine fields in refracting multipath environments that lead to the formation of caustics. This presentation illustrates the effectiveness of utilizing these methods to localize sources using propagation simulations in a refracting environment, and 200 to 300 Hz PhilSea10 propagation data (Worcester *et al.*, 2013) collected in a deep-ocean sound channel at source-array ranges of hundreds of kilometers. [Work supported by ONR.]

11:00

5aUW8. Block-sparse compressive localization for incipient tip vortex cavitation noise. MinSeuk Park, Woojae Seong (Underwater Acoust. Lab., Seoul Nat. Univ., 36-212,1, Gwanak-ro, Gwanak-gu, Seoul 08826, South Korea, mindal2@snu.ac.kr), Youngmin Choo (Sejong Univ., Seoul, South Korea), and Yongsung Park (Seoul Nat. Univ., Seoul, South Korea)

Marine propeller cavitation is a dominant noise source of ships. Thus, localizing the cavitation noise source is essential for subsequent remedy. Given that incipient tip vortex cavitation (TVC) noise radiates in all directions as a monopole source and a few noise sources exist in the vicinity of the propeller, the localization problem can be considered as a sparse signal reconstruction problem. Compressive sensing (CS) based localization technique utilizes a sparsity promoting constraint and solves the localization problem efficiently with high resolution. Block-sparse CS, based on block-sparsity, is adopted to process multiple frequency components of the sources coherently. Block-sparse CS localization of TVC shows superior performance with high resolution compared to the conventional CS based incoherent multiple frequency processing. To demonstrate the performance of the block-sparse CS localization, both synthetic and real cavitation tunnel experiment data are used.

11:15

5aUW9. Application of simulated annealing particle swarm optimization in underwater acoustic positioning optimization. JiangQiao Li (Systems Eng. Res. Inst. of CSSC, Beijing, China), Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn), Li Huangpu (Systems Eng. Res. Inst. of CSSC, Beijing, China), and Xianzhong Bu (Harbin Eng. Univ., Harbin, China)

In the calculation of underwater acoustic positioning, the errors of element position measurement, time delay measurement, attitude accuracy and sound velocity measurement will lead to the calculation errors of final positioning solution. The optimization method of solution is also one of the key technologies of location solution. The traditional statistical methods of solution mostly use arithmetic average optimization method and Newton iteration method. In this paper, the particle swarm optimization method based on simulated annealing is used to improve the estimation accuracy of location solution. The effectiveness of this method is verified by computer simulation and comparative analysis. First, the spherical intersection method of synchronous positioning and the hyperbolic intersection method of asynchronous positioning are introduced in this paper. Then, the method of measuring time delay and filtering signal is introduced. Finally, another factor affecting positioning accuracy is discussed. The method of solving nonlinear equation is discussed. By comparing the root mean square error of three methods under different signal-to-noise ratios, the signal-to-noise ratio is obtained. The accuracy of the three algorithms is almost the same when compared with each other, but at low SNR, the simulated annealing particle swarm optimization (SA-PSO) algorithm can calculate more accurate positioning solutions.

11:30

5aUW10. Passive underwater acoustic tags with curved symmetry for navigation and information encoding. Aprameya Satish, David Trivett, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30313, aprameya.satish@gatech.edu)

Passive acoustic beacons built of horizontally stratified materials have been designed in previous literature to assist in the navigation of autonomous underwater vehicles (UAV) equipped with SOund Navigation and Ranging (SONAR) instrumentation. These beacons reflect a characteristic acoustic signature which can be detected by the AUV as acoustic backscattering upon tag insonification. Currently, only backscattering from acoustic waves normally incident on a beacon can be detected by the AUV due to the beacon's planar geometry. To address this issue, this paper proposes the design of passive acoustic beacons with curved symmetry, whose acoustic backscattering is the same irrespective of the angle of source incidence. Simulation and experimental results are discussed for beacons made of concentric spherical shells as a proof of concept.

11:45

5aUW11. Measurement of arm length difference of interferometric fiber-optic hydrophone using extra-carrier modulation and interference fringe counting method. Jun Zhang, Yi Chen, Han Zhao, and Jiaheng Wang (Hangzhou Appl. Acoust. Res. Inst., No. 82, Guihuaxi Rd., Fuyang District, Hangzhou, Zhejiang 311400, China, 13957123130@139.com)

Due to the characteristics of low loss, passivity, and long-distance transmission, interferometric fiber-optic hydrophones and their arrays have been widely used in engineering detection applications. The arm length difference is one of its main performance parameters. A system for measuring of arm length difference using extra-carrier modulation and interference fringe counting method is introduced in detail. The propagation process of optical signal in the fiber-optic hydrophone is described. The mathematical expression of the phase of interferometric optical signal intensity caused by external carrier is given and analyzed. The expression of arm length difference and its evaluation of measurement uncertainty are obtained. Experimental study on measurement of arm length difference is developed. Both theoretical analysis and experimental results show that the product of the driving voltage of the fiber stretcher and the arm length difference has fixed value. Combining the known arm length difference value and the carrier driving voltage of the reference interferometer, the arm length difference of the hydrophone to be measured can be calculated with the value of its carrier driving voltage accurately. The uncertainty of measurement of the device and method described in this paper is better than 0.5%. The measuring range is 0.1 m to 500 m.

ETHICAL PRINCIPLES OF THE ACOUSTICAL SOCIETY OF AMERICA FOR RESEARCH INVOLVING HUMAN AND NON-HUMAN ANIMALS IN RESEARCH AND PUBLISHING AND PRESENTATIONS

The Acoustical Society of America (ASA) has endorsed the following ethical principles associated with the use of human and non-human animals in research, and for publishing and presentations. The principles endorsed by the Society follow the form of those adopted by the American Psychological Association (APA), along with excerpts borrowed from the Council for International Organizations of Medical Sciences (CIOMS). The ASA acknowledges the difficulty in making ethical judgments, but the ASA wishes to set minimum socially accepted ethical standards for publishing in its journals and presenting at its meetings. These Ethical Principles are based on the principle that the individual author or presenter bears the responsibility for the ethical conduct of their research and is publication or presentation.

Authors of manuscripts submitted for publication in a journal of the Acoustical Society of America or presenting a paper at a meeting of the Society are obligated to follow the ethical principles of the Society. Failure to accept the ethical principles of the ASA shall result in the immediate rejection of manuscripts and/or proposals for publication or presentation. False indications of having followed the Ethical Principles of the ASA may be brought to the Ethics and Grievances Committee of the ASA.

APPROVAL BY APPROPRIATE GOVERNING AUTHORITY

The ASA requires all authors to abide by the principles of ethical research as a prerequisite for participation in Society-wide activities (e.g., publication of papers, presentations at meetings, etc.). Furthermore, the Society endorses the view that all research involving human and non-human vertebrate animals requires approval by the appropriate governing authority (e.g., institutional review board [IRB], or institutional animal care and use committee [IACUC], Health Insurance Portability and Accountability Act [HIPAA], or by other governing authorities used in many countries) and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, then the intent of the ASA Ethical Principles described in this document must be met. All research involving the use of human or non-human animals must have met the ASA Ethical Principles prior to the materials being submitted to the ASA for publication or presentation.

USE OF HUMAN SUBJECTS IN RESEARCH-Applicable when human subjects are used in the research

Research involving the use of human subjects should have been approved by an existing appropriate governing authority (e.g., an institutional review board [IRB]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

Informed Consent

When obtaining informed consent from prospective participants in a research protocol that has been approved by the appropriate and responsible-governing body, authors must have clearly and simply specified to the participants beforehand:

1. The purpose of the research, the expected duration of the study, and all procedures that were to be used.
2. The right of participants to decline to participate and to withdraw from the research in question after participation began.
3. The foreseeable consequences of declining or withdrawing from a study.
4. Anticipated factors that may have influenced a prospective participant's willingness to participate in a research project, such as potential risks, discomfort, or adverse effects.
5. All prospective research benefits.
6. The limits of confidentially.
7. Incentives for participation.
8. Whom to contact for questions about the research and the rights of research participants. The office/person must have willingly provided an atmosphere in which prospective participants were able to ask questions and receive answers.

Authors conducting intervention research involving the use of experimental treatments must have clarified, for each prospective participant, the following issues at the outset of the research:

1. The experimental nature of the treatment;
2. The services that were or were not to be available to the control group(s) if appropriate;
3. The means by which assignment to treatment and control groups were made;
4. Available treatment alternatives if an individual did not wish to participate in the research or wished to withdraw once a study had begun; and
5. Compensation for expenses incurred as a result of participating in a study including, if appropriate, whether reimbursement from the participant or a third-party payer was sought.

Informed Consent for Recording Voices and Images in Research

Authors must have obtained informed consent from research participants prior to recording their voices or images for data collection unless:

1. The research consisted solely of naturalistic observations in public places, and it was not anticipated that the recording would be used in a manner that could have caused personal identification or harm, or
2. The research design included deception. If deceptive tactics were a necessary component of the research design, consent for the use of recordings was obtained during the debriefing session.

Client/Patient, Student, and Subordinate Research Participants

When authors conduct research with clients/patients, students, or subordinates as participants, they must have taken steps to protect the prospective participants from adverse consequences of declining or withdrawing from participation.

Dispensing With Informed Consent for Research

Authors may have dispensed with the requirement to obtain informed consent when:

1. It was reasonable to assume that the research protocol in question did not create distress or harm to the participant and involves:
   a. The study of normal educational practices, curricula, or classroom management methods that were conducted in educational settings
   b. Anonymous questionnaires, naturalistic observations, or archival research for which disclosure of responses would not place participants at risk of criminal or civil liability or damage their financial standing, employability, or reputation, and confidentiality
   c. The study of factors related to job or organization effectiveness conducted in organizational settings for which there was no risk to participants' employability, and confidentiality.
2. Dispensation is permitted by law.
3. The research involved the collection or study of existing data, documents, records, pathological specimens, or diagnostic specimens, if these sources are publicly available or if the information is recorded by the investigator in such a manner that subjects cannot be identified, directly or through identifiers linked to the subjects.

Offering Inducements for Research Participation

(a) Authors must not have made excessive or inappropriate financial or other inducements for research participation when such inducements are likely to coerce participation.

(b) When offering professional services as an inducement for research participation, authors must have clarified the nature of the services, as well as the risks, obligations, and limitations.

Deception in Research

(a) Authors must not have conducted a study involving deception unless they had determined that the use of deceptive techniques was justified by the study's significant prospective scientific, educational, or applied value and that effective non-deceptive alternative procedures were not feasible.

(b) Authors must not have deceived prospective participants about research that is reasonably expected to cause physical pain or severe emotional distress.

(c) Authors must have explained any deception that was an integral feature of the design and conduct of an experiment to participants as early as was feasible, preferably at the conclusion of their participation, but no later than at the conclusion of the data collection period, and participants were freely permitted to withdraw their data.

Debriefing

(a) Authors must have provided a prompt opportunity for participants to obtain appropriate information about the nature, results, and conclusions of the research project for which they were a part, and they must have taken reasonable steps to correct any misconceptions that participants may have had of which the experimenters were aware.

(b) If scientific or humane values justified delaying or withholding relevant information, authors must have taken reasonable measures to reduce the risk of harm.

(c) If authors were aware that research procedures had harmed a participant, they must have taken reasonable steps to have minimized the harm.

HUMANE CARE AND USE OF NON-HUMAN VERTEBRATE ANIMALS IN RESEARCH-Applicable when non-human vertebrate animals are used in the research

The advancement of science and the development of improved means to protect the health and well being both of human and non-human vertebrate animals often require the use of intact individuals representing a wide variety of species in experiments designed to address reasonable scientific questions. Vertebrate animal experiments should have been undertaken only after due consideration of the relevance for health, conservation, and the advancement of scientific knowledge. (Modified from the Council for International Organizations of Medical Science (CIOMS) document: "International Guiding Principles for Biomedical Research Involving Animals1985"). Research involving the use of vertebrate animals should have been approved by an existing appropriate governing authority (e.g., an institutional animal care and use committee [IACUC]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

The proper and humane treatment of vertebrate animals in research demands that investigators:

1. Acquired, cared for, used, interacted with, observed, and disposed of animals in compliance with all current federal, state, and local laws and regulations, and with professional standards.

2. Are knowledgeable of applicable research methods and are experienced in the care of laboratory animals, supervised all procedures involving animals, and assumed responsibility for the comfort, health, and humane treatment of experimental animals under all circumstances.

3. Have insured that the current research is not repetitive of previously published work.

4. Should have used alternatives (e.g., mathematical models, computer simulations, etc.) when possible and reasonable.

5. Must have performed surgical procedures that were under appropriate anesthesia and followed techniques that avoided infection and minimized pain during and after surgery.

6. Have ensured that all subordinates who use animals as a part of their employment or education received instruction in research methods and in the care, maintenance, and handling of the species that were used, commensurate with the nature of their role as a member of the research team.

7. Must have made all reasonable efforts to minimize the number of vertebrate animals used, the discomfort, the illness, and the pain of all animal subjects.

8. Must have made all reasonable efforts to minimize any harm to the environment necessary for the safety and well being of animals that were observed or may have been affective as part of a research study.

9. Must have made all reasonable efforts to have monitored and then mitigated any possible adverse affects to animals that were observed as a function of the experimental protocol.

10. Who have used a procedure subjecting animals to pain, stress, or privation may have done so only when an alternative procedure was unavailable; the goal was justified by its prospective scientific, educational, or applied value; and the protocol had been approved by an appropriate review board.

11. Proceeded rapidly to humanely terminate an animal's life when it was necessary and appropriate, always minimizing pain and always in accordance with accepted procedures as determined by an appropriate review board.

PUBLICATION and PRESENTATION ETHICS-For publications in ASA journals and presentations at ASA sponsored meetings

Plagiarism

Authors must not have presented portions of another's work or data as their own under any circumstances.

Publication Credit

Authors have taken responsibility and credit, including authorship credit, only for work they have actually performed or to which they have substantially contributed. Principal authorship and other publication credits accurately reflect the relative scientific or professional contributions of the individuals involved, regardless of their relative status. Mere possession of an institutional position, such as a department chair, does not justify authorship credit. Minor contributions to the research or to the writing of the paper should have been acknowledged appropriately, such as in footnotes or in an introductory statement.

Duplicate Publication of Data

Authors did not publish, as original data, findings that have been previously published. This does not preclude the republication of data when they are accompanied by proper acknowledgment as defined by the publication policies of the ASA.

Reporting Research Results

If authors discover significant errors in published data, reasonable steps must be made in as timely a manner as possible to rectify such errors. Errors can be rectified by a correction, retraction, erratum, or other appropriate publication means.

DISCLOSURE OF CONFLICTS OF INTEREST

If the publication or presentation of the work could directly benefit the author(s), especially financially, then the author(s) must disclose the nature of the conflict:

1) The complete affiliation(s) of each author and sources of funding for the published or presented research should be clearly described in the paper or publication abstract.

2) If the publication or presentation of the research would directly lead to the financial gain of the authors(s), then a statement to this effect must appear in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper.

3) If the research that is to be published or presented is in a controversial area and the publication or presentation presents only one view in regard to the controversy, then the existence of the controversy and this view must be provided in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper. It is the responsibility of the author to determine if the paper or presentation is in a controversial area and if the person is expressing a singular view regarding the controversy.

Sustaining Members of the Acoustical Society of America

The Acoustical Society is grateful for the financial assistance being given by the Sustaining Members listed below and invites applications for sustaining membership from other individuals or corporations who are interested in the welfare of the Society.

Application for membership may be made to the Executive Director of the Society and is subject to the approval of the Executive Council. Dues of $1000.00 for small businesses (annual gross below $100 million) and $2000.00 for large businesses (annual gross above $100 million or staff of commensurate size) include a subscription to the *Journal* as well as a yearly membership certificate suitable for framing. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year.

Additional information and application forms may be obtained from Elaine Moran, Office Manager, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: elaine@acousticalsociety.org

Acentech Incorporated
www.acentech.com
Cambridge, Massachusetts
Consultants in Acoustics, Audiovisual and Vibration

ACO Pacific Inc.
www.acopacific.com
Belmont, California
Measurement Microphones, the ACOustic Interface™ System

Acoustics First Corporation
www.acousticsfirst.com
Richmond, Virginia
Materials to Control Sound and Eliminate Noise™

American Institute of Physics
www.aip.org
College Park, Maryland
Career resources, undergraduate education, science policy, and history

BBN Technologies
www.bbn.com
Cambridge, Massachusetts
R&D company providing custom advanced research based solutions

G.R.A.S.
Sound & Vibration ApS
www.gras.dk
Holte, Denmark
Measurement microphones, Intensity probes, Calibrators

Kinetics Noise Control, Inc.
www.kineticsnoise.com
Dublin, Ohio
Kinetics manufactures products to address vibration and noise control, room acoustics, and seismic restraint concerns for almost any building application

Knowles Electronics, Inc.
www.knowles.com
Itasca, Illinois
Manufacturing Engineers: Microphones, Recording, and Special Audio Products

Massa Products Corporation
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

National Gypsum Company
www.nationalgypsum.com
Charlotte, North Carolina
Manufacturer of acoustically enhanced gypsum board

Raytheon Company
Integrated Defense Systems
www.raytheon.com
Portsmouth, Rhode Island
Sonar Systems and Oceanographic Instrumentation: R&D in Underwater Sound Propagation and Signal Processing

ROXUL, Inc. – Core Solutions (OEM)
www.roxul.com
Milton, ON, Canada
Offers a variety of insulation products ranging in density and dimension to meet any production requirements. Products are successfully used in numerous acoustical OEM applications providing solutions for a number of industries

Thales Underwater Systems
www.thales-naval.com
Somerset, United Kingdom
Prime contract management, customer support services, sonar design and production, masts and communications systems design and production

3M Personal Safety Division (PSD)
www.3m.com/occsafety
Minneapolis, Minnesota
Products for personal and environmental safety, featuring E·A·R and Peltor brand hearing protection and fit testing, Quest measurement instrumentation, audiological devices, materials for control of noise, vibration, and mechanical energy, and the E·A·RCALSM laboratory for research, development, and education, NVLAP-accredited since 1992.
Hearing conservation resource center
www.e-a-r.com/hearingconservation

Wenger Corporation
www.wengercorp.com
Owatonna, Minnesota
Design and Manufacturing of Architectural Acoustical Products including Absorbers, Diffusers, Modular Sound Isolating Practice Rooms, Acoustical Shells and Clouds for Music Rehearsal and Performance Spaces

Wyle Laboratories
www.wyle.com
Arlington, Virginia
The Wyle Acoustics Group provides a wide range of professional services focused on acoustics, vibration, and their allied technologies, including services to the aviation industry

ACOUSTICAL · SOCIETY · OF · AMERICA

APPLICATION FOR SUSTAINING MEMBERSHIP

The Bylaws provide that any person, corporation, or organization contributing annual dues as fixed by the Executive Council shall be eligible for election to Sustaining Membership in the Society.

Dues have been fixed by the Executive Council as follows: $1000 for small businesses (annual gross below $100 million); $2000 for large businesses (annual gross above $100 million or staff of commensurate size). Dues include one year subscription to *The Journal of the Acoustical Society of America* and programs of Meetings of the Society. Please do not send dues with application. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year. If elected, you will be billed.

Name of Company _____

Address _____

Telephone: _____ Fax: _____

E-mail: _____ WWW: _____

Size of Business: ☐ Small business ☐ Small business—No Journal ☐ Large business

Type of Business _____

Please enclose a copy of your organization's brochure.

In listing of Sustaining Members in the Journal and on the ASA homepage we should like to indicate our products or services as follows:

_____

(please do not exceed fifty characters)

Name of company representative to whom journal should be sent:

_____

It is understood that a Sustaining Member will not use the membership for promotional purposes.

Signature of company representatives making application:

_____

Please send completed applications to: Executive Director, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, (516) 576-2360, asa@acousticalsociety.org

MEMBERSHIP INFORMATION AND APPLICATION INSTRUCTIONS

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of the application; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

BENEFITS OF MEMBERSHIP	Full Member	Associate	ce-Associate	Student
JASA Online–Vol. 1 (1929) to present	*	*	*	*
JASA tables of contents e-mail alerts	*	*	*	*
JASA, printed or CD ROM	*	*		
JASA Express Letters–online	*	*	*	*
Acoustics Today–the quarterly magazine	*	*	*	*
Proceedings of Meetings on Acoustics	*	*	*	*
Noise Control and *Sound*, *It's Uses and Control*–online archival magazines	*	*	*	*
Acoustics Research Letters Online (ARLO)–online archive	*	*	*	*
Programs for Meetings	Online	Online	Online	Online
Meeting Calls for Papers	Online	Online	Online	Online
Reduced Meeting Registration Fees	*	*		*
5 Free ASA standards per year-download only	*	*		*
Standards Discounts	*	*		*
Society Membership Directory	Online	Online	Online	Online
Electronic Announcements	*	*	*	*
Physics Today	*	*	*	*
Eligibility to vote and hold office in ASA	*			
Eligibility to be elected Fellow	*	*		
Participation in ASA Committees	*	*	*	*

QUALIFICATIONS FOR EACH GRADE OF MEMBERSHIP AND ANNUAL DUES

Student: Any student interested in acoustics who is enrolled in an accredited college or university for half time or more (at least eight semester hours). Dues: $45 per year.
Associate: Any individual interested in acoustics. Dues: $95 per year. After five years, the dues of an Associate increase to that of a full Member.
Corresponding Electronic Associate: Any individual residing in a developing country who wishes to have access to ASA's online publications only including *The Journal of the Acoustical Society of America* and Meeting Programs [see http://acousticalsociety.org/membership/membership_and_benefits]. Dues $45 per year.
Member: Any person active in acoustics, who has an academic degree in acoustics or in a closely related field or who has had the equivalent of an academic degree in scientific or professional experience in acoustics, shall be eligible for election to Membership in the Society. A nonmember applying for full Membership will automatically be made an interim Associate Member, and must submit $95 with the application for the first year's dues. Election to full Membership may require six months or more for processing; dues as a full Member will be billed for subsequent years.

JOURNAL OPTIONS AND COSTS FOR FULL MEMBERS AND ASSOCIATE MEMBERS ONLY

- ONLINE JOURNAL. All members will receive access to the *The Journal of the Acoustical Society of America (JASA)* at no charge in addition to dues.
- PRINT JOURNAL. Twelve monthly issues of *The Journal of the Acoustical Society of America*. **Cost: $35 in addition to dues.**
- CD-ROM. The CD ROM mailed bimonthly. This option includes all of the material published in the Journal on CD ROM. **Cost: $35 in addition to dues.**
- COMBINATION OF THE CD-ROM AND PRINTED JOURNAL. The CD-ROM mailed bimonthly and the printed journal mailed monthly. **Cost: $70 in addition to dues.**
- **EFFECTIVE DATE OF MEMBERSHIP**. If your application for membership and dues payment are received by 15 September, your membership and Journal subscription will begin during the current year and you will receive all back issues for the year. If you select the print journal option. If your application is received after 15 September, however, your dues payment will be applied to the following year and your Journal subscription will begin the following year.

OVERSEAS AIR DELIVERY OF JOURNALS

Members outside North, South, and Central America can choose to have print journals sent by air freight at a cost of $165 in addition to dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

ACOUSTICAL SOCIETY OF AMERICA

1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, asa@acousticalsociety.org

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Dues Rcvd _____
Aprvd by Ed _____
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)

CHECK ONE BOX IN EACH COLUMN ON THE RIGHT

☐ NON-MEMBER APPLYING FOR:
☐ MEMBER REQUESTING TRANSFER TO:

☐ STUDENT MEMBERSHIP
☐ ASSOCIATE MEMBERSHIP
☐ CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP
☐ FULL MEMBERSHIP

Note that your choice of journal option *may* increase or decrease the amount you must remit.

SELECT JOURNAL OPTION:

Student members will automatically receive access to The Journal of the Acoustical Society of America online at no charge in addition to dues. Remit $45. (Note: Student members may also receive the Journal on CD ROM at an additional charge of $35.)

Corresponding Electronic Associate Members will automatically receive access to The Journal of the Acoustical Society of America and Meeting Programs online at no charge in addition to dues. Remit $45.

Applicants for **Associate or full Membership** must select one Journal option from those listed below. Note that your selection of journal option determines the amount you must remit.

[ ] Online access only—$95
[ ] Online access plus print Journal $130
[ ] Online access plus CD ROM—$130
[ ] Online access plus print Journal and CD ROM combination—$165

Applications received after 15 September: Membership and Journal subscriptions begin the following year.

OPTIONAL AIR DELIVERY: Applicants from outside North, South, and Central America may choose air freight delivery of print journals for an additional charge of $165. If you wish to receive journals by air, remit the additional amount owed with your dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

LAST NAME	FIRST NAME	MIDDLE INITIAL	MS/MR/MRS/DR/PROF
HOME ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
NAME OF ORGANIZATION OR BUSINESS			
DEPARTMENT			
ORGANIZATION ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY

BUSINESS TELEPHONE: AREA CODE/NUMBER	MOBILE PHONE: AREA CODE/NUMBER	HOME TELEPHONE: AREA CODE/NUMBER
E-MAIL ADDRESS: (PRINT CLEARLY)		
DATE AND PLACE OF BIRTH (Req'd for Awards and Emeritus Status)	SEX: ☐ Female ☐ Male	

HIGHEST ACADEMIC DEGREE	MONTH/YEAR	FIELD	INSTITUTION GRANTING DEGREE
OTHER DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE

CHECK PERFERRED ADDRESS FOR MAIL: ☐ HOME ☐ ORGANIZATION

Part I Continued ➔

PART I CONTINUED: ACOUSTICAL AREAS OF INTEREST TO APPLICANT. Indicate your three main areas of interest below, using 1 for your main interest, 2 for your second, and 3 for your third interest. (DO NOT USE CHECK MARKS.)

☐ ACOUSTICAL OCEANOGRAPHY **M**
☐ ANIMAL BIOACOUSTICS **L**
☐ ARCHITECTURAL ACOUSTICS **A**
☐ BIOMEDICAL ACOUSTICS **K**
☐ COMPUTATIONAL ACOUSTICS **O**
☐ ENGINEERING ACOUSTICS **B**
☐ MUSICAL ACOUSTICS **C**
☐ NOISE & NOISE CONTROL **D**
☐ PHYSICAL ACOUSTICS **E**
☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
☐ SIGNAL PROCESSING IN ACOUSTICS **N**
☐ SPEECH COMMUNICATION **H**
☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
☐ UNDERWATER ACOUSTICS **J**

PART II: APPLICATION FOR STUDENT MEMBERSHIP

NAME AND ADDRESS OF COLLEGE OR UNIVERSITY WHERE PRESENTLY ENROLLED		
DEGREE EXPECTED	MONTH & YEAR DEGREE EXPECTED	NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER
PRINT NAMES & E-MAIL ADDRESSES OF TWO FACULTY MEMBERS CERTIFYING THAT YOU ARE REGISTERED FOR AT LEAST ONE-HALF OF FULL TIME		
SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME		
SIGNATURE OF APPLICANT		DATE

PART III: APPLICATION FOR ASSOCIATE MEMBERSHIP, CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP OR FULL MEMBERSHIP (and interim Associate Membership)

SUMMARIZE YOUR MAJOR PROFESSIONAL EXPERIENCE on the lines below: list employers, duties and position titles, and dates, beginning with your present position. Attach additional sheets if more space is required.

CONTRIBUTIONS TO ACOUSTICS: LIST MAIN PUBLICATIONS, PATENTS, ETC. Attach separate sheets if required.

SPONSORS AND REFERENCES: An application for full Membership requires the names, addresses, and signatures of two references who must be **full Members or Fellows** of the Acoustical Society. Names and signatures are NOT required for Associate Membership, Corresponding Electronic Associate Membership or Student Membership applications.

PRINT NAME OF REFERENCE (required for Full Member applications only)	PRINT NAME OF SECOND REFERENCE (required for Full Member applications only)
ADDRESS OF REFERENCE	ADDRESS OF SECOND REFERENCE
SIGNATURE OF REFERENCE (required for Full Member applications only)	SIGNATURE OF SECOND REFERENCE (required for Full Member applications only)
SIGNATURE OF APPLICANT	DATE

MAIL THIS COMPLETED APPLICATION, WITH APPROPRIATE PAYMENT TO: ACOUSTICAL SOCIETY OF AMERICA, 1305 WALT WHITMAN ROAD, SUITE 300, MELVILLE, NY 11747-4300.

METHOD OF PAYMENT

☐ Check or money order enclosed for $ _____ (U.S. funds/drawn on U.S. bank)

☐ American Express ☐ VISA ☐ MasterCard Signature _____

(Credit card orders must be signed)

Account Number ☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐

Expiration Date Mo. ☐☐ Yr. ☐☐

Security Code ☐☐☐☐

Due to security risks and Payment Card Industry (PCI) data security standards e-mail is NOT an acceptable way to transmit credit card information. Please return this form by Fax (631-923-2875) or by postal mail.

Regional Chapters and Student Chapters

Anyone interested in becoming a member of a regional chapter or in learning if a meeting of the chapter will be held while he/she is in the local area of the chapter, either permanently or on travel, is welcome to contact the appropriate chapter representative. Contact information is listed below for each chapter representative.

Anyone interested in organizing a regional chapter in an area not covered by any of the chapters below is invited to contact the Cochairs of the Committee on Regional Chapters for information and assistance: Evelyn Hoglund, Ohio State University, hoglund1@osu.edu and Kenneth W. Good, Jr., Armstrong World Industries, Inc., kwgoodjr@armstrong.com

AUSTIN STUDENT CHAPTER
Benjamin C. Treweek
Austin, TX
austinacousticalsociety@gmail.com

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to papers presented at

177th Meeting: Acoustical Society of America

Abavisani, Ali–1663
Abawi, Ahmad T.–1692
Abboud, Monica–1922
Abdelaziz, Ahmed–1698
Abdelrahman, Anas M.–1881
Abdulmagid, Zaineb–1902
Abenojar, Eric C.–1892
Abrams, Kristina S.–1879
Abu El Adas, Sandy–1690
Acharya, Mukunda–1804
Acosta Carrasco, Carlos–1876
Adabi, Saba–1891
Adhikari, Kaushallya–1733
Afrough, Mohammad–Cochair Session 1aSAb (1667)
Agarwal, Anil–1921
Ahlstrom, Jayne B.–1718, 1821, 1822
Ahnert, Wolfgang–1854
Ahrens, Axel–1871, Cochair Session 1aPP (1663), Cochair Session 4aPP (1871)
Ahrens, Jens–1818, 1889
Ahuja, Krish–1902
Aichinger, Philipp–1909
Akahane-Yamada, Reiko–1826
Akchurin, Adel–1769
Aker, Scott–1880
Akita, Takeshi–1867
Alamatsaz, Nima–1759
Alamri, Sarah–1828
Alberts, W. C. Kirkpatrick–1662, 1681, Cochair Session 1aPA (1660), Cochair Session 1pPAa (1680)
Alexander, Joshua M.–1875
Alexandre, David–1742, 1743
Alhussaini, Mohamed–1717
Ali, Mir H.–1674
Aliabouzar, Mitra–1703
Aliverti, Andrea–1656
Alizad, Azra–1891, 1893
Allen, Jont–1663, 1797
Allen, Jont b.–1763
Allen, Makenzie B.–1936
AlMakadma, Hammam–1705
Almogbil, Hanaa H.–1894
Alter, Ross E.–1870, 1901
Altman, Michael–1862
Alu, Andrea–1761
Alù, Andrea–1761
Álvarez, Fernando J.–1884
Alvarez, Ivette–1817
Alvord, David–1902
Amaral, Jennifer–1671
Amburgey, Kassidy N.–1879
An, Buchao–1693
Anand, Ishan–1704
Anand, Supraja–1797
Anastasio, Mark–1858
Anderson, Brian E.–1704, 1715, 1755, 1925
Anderson, Elizabeth–1879
Anderson, Jennifer L.–1702, 1746
Anderson, Ryan–1720
Anderson, Samira B.–1705, 1908
Anderson, Stephan–1726, 1896
Anderson, Victoria R.–1918
Andersson, Carl–1818, 1889
Andre, Michael P.–1921
Andrews, Peter A.–1931
Annand, Colin–1799
Anthony, Brian–1922
Anthony, Gregory–1747
Aoyama, Katsura–1826
Aparicio, Joaquín–1884
Aquino, Wilkins–1891
Araujo, Carlos–1902
Ardhuin, Fabrice–1868
Arenberg, Julie G.–1905
Aristizabal, Orlando–1701
Arjmandi, Meisam K.–1690, 1766
Arlin, Jennifer B.–1932
Arnal, Bastien–1779
Arnoult, Jennifer–1797
Arretche, Ignacio–1760
Arrieta, Rodolfo–1691
Arts, Floor–1821
Arvanitis, Costas–1859
Aspöck, Lukas–1889
Assink, Jelle D.–1870
Assous, Said–1769, Cochair Session 2pSP (1768)
Athanassiadis, Athanasios G.–1655
Atwood, Stephanie–1766
Au, Whitlow–1774
Averbuch, Gil–1868
Azman, Amanda S.–1866
Babaniyi, Olalekan–1891
Babaniyi, Olalekan A.–1858
Babel, Molly E.–1789
Badel, Gloria-Tabea–1793
Bader, Kenneth B.–1746, 1747
Bader, Rolf–1676, 1709, 1783
Badiey, Mohsen–1803, 1935, 1936, Chair Session 3aUW (1803)
Bae, Myungjin–1865, 1896, 1921, 1922
Bae, Seonggeon–1921
Baek, Hyunah–1826, 1926
Baese-Berk, Melissa M.–1789, 1827
Baglione, Melody–1883
Bahn, Curtis–1784
Baik, Kyungmin–1935
Bailey, Jhonelle–1731
Bailey, Peter J.–1909
Bainbridge, Constance M.–1766
Baker, Steven R.–1682
Baldinelli, Natalie–1798
Ball, Natalie J.–1878
Ballard, Megan S.–1672, 1803, 1887
Baltzell, Lucas–1720
Bamber, Jeffrey C.–1777
Banks, H. T.–1781
Banks, Jacob–1872
Barach, Paul–1658
Barber, Adrian–1817
Barbone, Paul E.–1858, 1891
Barclay, David R.–1654
Bargatin, Igor–1760
Barker, Daniel–1767
Barnard, Andrew R.–1883
Barros, Abner C.–1802
Bartholmai, Brian–1673
Baskent, Deniz–1796, 1821
Batson, Jesse–1860
Bayly, Philip–1786
Baz, Amr M.–1686
Beauchamp, James W.–1709
Beaudry, Lambert–1928
Beck, Benjamin–1665, 1725, 1824, Cochair Session 3pSA (1823)
Becker, Julian–1898
Beddor, Patrice S.–1911
Behboudi, Ramin–1658
Behrens, Tobias–1854
Belcourt, Josee–1801
Benders, Titia–1767
Benedett, Sharon–1826
Benjamin, Alex–1922
Bent, Tessa–1798, 1819, 1910
Benton, Rachel P.–1704
Berdahl, Edgar J.–1709, 1710
Berg, Katelyn A.–1796
Bergelson, Elika–1729, 1763
Bergeson, Tonya–1729
Beristain, Sergio–1652
Berkson, Kelly–1928
Berman, Alycia G.–1702, 1746
Bernstein, Joshua G. W.–1721
Bernstein, Leslie R.–1684
Berry, Alain–1896
Bertenthal, Bennett I.–1764
Bertolino, Angela–1733
Best, Virginia–1720, 1722, 1759
Betchkal, Davyd–1679
Betts, Abigail–1765
Bharadwaj, Hari M.–1663, 1717, 1718, 1872, 1906
Bharatula, Lakshmi Deepika–1745, 1894
Bhardwaj, Ananya–1742, 1743
Bieszczad, Kasia M.–1905
Biggs, Christopher–1807, Chair Session 3pAB (1806)
BinKhamis, Ghada–1907
Bishop, Brian B.–1661
Bishop, Jordan W.–1868
Bissell, Marie–1929
Biswas, Amitava–1765, 1775, 1878
Bittner, James–1756
Bixler, Tiffany A.–1918
Blackstock, David T.–1713
Blackwell, John–1674
Blanc-Benon, Philippe–1885
Blanford, Thomas E.–1771, 1935
Bliefnick, Jay–1658, Chair Session 1aNS (1657)
Bliss, Donald B.–1889
Blom, Philip–1901, Cochair Session 4aPA (1867), Cochair Session 4pPAa (1900)
Blotter, Jonathan–1704, 1712, 1791, 1896
Blottman, John B.–1863
Blyth, Matthew–1768
Boatti, Elisa–1719
Bohnert, Amanda–1928
Bollen, Viktor–1746, 1747
Bonnel, Julien–1672, 1801
Borden, Mark–1892
Borjigan, Agudemu–1872
Born, Alexander C.–1711
Bortoli, Fabio D.–1668
Boschi, Valerio–1668
Bosen, Adam K.–1821
Bosker, Hans Rutger–1788
Bossy, Emmanuel–1779
Botero, Natalia–1804
Botta, Jeanine–1678, Cochair Session 1pNS (1678)
Bouavichith, Dominique A.–1911, Cochair Session 3pPPa (1819)
Boucher, Matthew–1888, 1899
Boyce, Suzanne–1798, 1799
Boyce, Suzanne E.–1794
Boyce, Suzanne–1799
Boyden, Andrew–1883
Bozorg, Narjes–1932
Braasch, Jonas–1708, 1737, 1780, 1784, 1814, 1873, 1874, 1875, Cochair Session 3aMU (1782), Cochair Session 3pMU (1813)
Bradlow, Ann–1749
Bramhall, Naomi–1907
Brandewie, Eugene–1875
Bray, Wade–1917
Brennan, Marc–1878
Bressler, Scott–1722
Breton, Daniel J.–1800
Bridgewater, Benjamin–Cochair Session 3pAA (1805)
Brill, Laura C.–1780
Brimijoin, Owen–1871
Brinkworth, Russell–1864
Brooks, Bennett M.–1696, Cochair Session 2pNS (1752)
Brooks, Todd–1708
Broselow, Ellen–1826
Brown, Daniel C.–1770, 1771, 1935
Brown, Elisabeth M.–1787, 1800
Brown, Emma–1798
Brown, Kelly–1876
Browning, David–1807
Browning, Jenna–1904
Brownstead, Cale F.–1770, 1771
Bruce, Madeleine–1764
Brunet, Thomas–1762
Brungart, Douglas–1879
Brungart, Douglas S.–1721
Brungart, Timothy A.–1794
Bu, Xianzhong–1693, 1732, 1936

Buchwald, Adam–1933
Buckingham, Michael J.–1654
Bulla, Wesley–1877
Bunce, John–1763
Burch, Katrina–1901
Burkard, Robert F.–1856
Burke, Kali–1705, 1718
Burkinshaw, Kelly D.–1766
Burleson, Andrew–1873
Busch-Vishniac, Ilene–1923
Buskohl, Philip–1668, 1686
Buss, Emily–1684, 1721, 1790, 1879, 1904
Butko, Daniel–1805
Butler, Brooks A.–1816
Butler, Elizabeth–1909
Butts, Carla–1901
Byers-Heinlein, Krista–1728, 1766
Byram, Brett–1702
Byrd, Dani–1926
Byrick, Wilson–1711
Byrne, Andrew–1806, 1879
Byrne, David C.–1815
Byrne, Matthew S.–1761
Cai, Huaizhen–1888
Cai, Ruofan–1765
Cain, Charles A.–1862
Calloway, Ian–1932
Calloway, Ian C.–1911
Calvarese, Russell–1883
Calvert, Emma Kate–1910
Campanella, Sarah–1764
Campbell, Adam R.–1816
Campbell, Linda–1767
Campbell, Meghan R.–1746
Campbell, Tyler–1787, 1817
Cannon, Shauntelle–1820
Cariani, Peter–1785, 1813
Carlson, Dan–1775
Carlson, Thomas A.–1813
Carlyon, Robert P.–1721
Carney, Laurel H.–1664, 1685, 1719, 1758
Carrascal, Carolina A.–1922
Carriere, Madison–1861
Carroll, Yulia–1866
Carter, Yaminah D.–1915
Casacchia, Alicia–1811
Case, Jade–1709
Casillas, Marisa–1729, 1763
Centner, Connor S.–1894
Champagne, Matthew J.–1932
Chandra, Kavitha–1733, 1817, 1864
Chandrasekaran, Bharath–1820
Chandrasekaran, Sri Nivas–1771
Charles, Sherman D.–1795
Chatterjee, Monita–1820
Chatterjee Bhowmick, Diti–1894
Chatziioannou, Vasileios–1677, Chair Session 1pMU (1676)
Chavali, Vaibhav–1733
Chavez, Craig–1925
Chee, Adrian J.–1701, 1703
Chen, Fei–1879
Chen, Hao–1768
Chen, Hong–1859, 1862, Cochair Session 4aBAb (1860)
Chen, Jessica–1717, 1723
Chen, John J.–1675
Chen, Johnny L.–1672, 1920
Chen, Melinda–1922
Chen, Pei-Tai–1881
Chen, Qiyang–1703, 1859
Chen, Shuwen–1825
Chen, Si–1825, 1929
Chen, Vincent W.–1686
Chen, Xiaohua–1769
Chen, Yanyu–1687
Chen, Yao–1862
Chen, Yi–1937
Chen, Yuxiao–1659
Cheng, Andrew–1825
Cheng, Hung-Shao–1933
Cheng, Ruoqian–1828
Cheng, Zefeng–1802
Cheyne, Stanley A.–1923
Chhabra, Manish–1895
Chien, Yu-Fu–1913
Childs, Carter J.–1895
Chin-Bing, Stanley A.–1748
Chitale, Kedar C.–1787, 1817, Cochair Session 3aPA (1786), Cochair Session 3pPA (1817)
Chitnis, Parag V.–1778, 1811
Cho, Sungbeen–1880
Chodroff, Eleanor–1689, 1933
Choi, Hwiyong–1883
Choi, Inyong–1717
Choi, Yongki–1894
Choo, Youngmin–1936
Christian, Andrew–1685, 1899
Christoph, Marcus–1661
Chung, Andy–1753, 1918
Chunqi, Li–1810
Cibelli, Emily–1798
Clark, Cathy Ann–1804
Clark, Kelli–1722, 1874
Clauss, Michael–1872
Clayards, Meghan–1913
Clopper, Cynthia G.–1910, 1930
Clot, Arnau–1792
Cloud, Steven–1765
Coburn, Jeremy–1928
Coffeen, Robert C.–1697
Cohen, Joshua–1894
Coiado, Olivia–1923
Coila, Andres–1715
Colaianni, Julie–1777
Colburn, H. Steven–1759, 1813
Cole, Cliston–1797
Cole, Jennifer–1933
Coleman, John P.–1762
Coletta, Louise–1810
Collaboration, Infant-Directed Vocalizations–1766
Collier, Sandra L.–1662
Colosi, John A.–1653, 1803, Cochair Session 1aAO (1653)
Combs, Aleah D.–1910, 1931
Conk, Matthew–1741
Conklin, Jenna–1827, 1929
Conlon, Stephen–1791
Connors, Brady–1922
Conroy, Christopher–1683, 1719, Cochair Session 1pPP (1683)
Cook, Mylan R.–1816, 1870
Cooper, Jennifer–Cochair Session 3aCA (1780), Cochair Session 4aUW (1885)
Copetti, Roberto–1656
Cormack, John M.–1713, 1755, 1756
Corry, Brian–1739, 1740, Cochair Session 2pAA (1738)
Costley, R. D.–1660
Cottingham, James P.–1676, 1709
Cox, Michael T.–1862, 1891
Cox, Nathan–1742
Coxson, Gregory E.–1700
Coyle, Whitney L.–1677, Chair Session 2pMU (1750)
Coyne, Robert–1892
Craft, Justin T.–1911
Cristia, Alex–1729
Crouzet, Olivier–1914
Cuervo, Sisan–1798
Cuff, Nicole–1695
Cui, Huachen–1761
Cui, Shichao–1725
Cull, Molly–1767
Culnan, John–1827, 1911
Culver, R. Lee–Cochair Session 2pSP (1768)
Cummer, Steven–1666, 1687, 1727, 1761, 1762, 1882
Curtin, Suzanne–1766
Cushing, Colby W.–1727
Cybulski, Luke–1858
Cychosz, Margaret–1927
Dahlbom, David A.–1784, 1814
Dai, Huanping–1684
Damen, Frederick W.–1702
Daniel, Timothy D.–1692
Dannemann, Fransiska K.–1901
Darabi, Amir–1725
Darcy Mahoney, Ashley–1657
daSilva, Elizabeth B.–1764
Daszynski, Camille–1894
Dau, Torsten–1873
Daum, Moritz–1766
Davies-Venn, Evelyn E.–1664, 1879
Davies, Patricia–Cochair Session 4pNS (1897)
Davis, Eric S.–1925
Davis, Joshua S.–1691
Davis, Matthew–1723
Davis, Rebecca F.–1827
Davis, Rebekah–1860
Dawant, Benoit–1796
Dayavansha, E. G. Sunethra–1862
Dean, Cleon E.–1903
Deane, Grant B.–1654, 1887
De Carlo, Marine–1868
Degertekin, F. Levent–1881
De Gregorio, Chiara–1919
Dei, Kaz–1702
de Jong, Kenneth–1928
De Leon, Al C.–1892
Delfarah, Masood–1874
Dellwo, Volker–1766, 1929
Deloche, François–1721
DeLong, Caroline M.–1856
Demi, Libertario–1674, Cochair Session 1aBAb (1656), Cochair Session 1pBA (1673)
Deng, Yuanchen–1762
Denis, Max–1662, 1893, Cochair Session 3aPA (1786), Cochair Session 3pPA (1817)
Denison, Michael–1715
Dent, Micheal L.–1718, 1806
De Palma, Paul–1729, 1764
Derleth, R. Peter–1720
Desmet, Wim–1888
Dessi-Olive, Jonathan–1855
Dettmer, Jan–1801
Devries, Lindsay A.–1905
Di, Xiao–1661
Dickey, Darryl–1902
Dickey, Emily–1909
Dieckman, Eric A.–1707, 1781, 1864
Diedesch, Anna C.–Cochair Session 4pPP (1904)
Dietz, Mathias–1759
Digerness, Joseph–1738
Dilley, Laura–1690, 1730, 1766, Cochair Session 2aSCb (1730), Cochair Session 2pSC (1763)
DiMarzio, Charles A.–1810
DiNino, Mishaela–1905
Dittberner, Andrew–1875, 1876, 1900
Dixit, Omkar–1722
Dlubac, James J.–1823
Dmitrieva, Olga–1827, 1929, Chair Session 5aSC (1926)
Doan, Hung–1925
Dodds, Peter–1658
Dodsworth, Robin M.–1931
Doebler, William–1899
Doedens, Ric–1652
Doggett, Felicia–1710
Dolan, Sophia E.–1784
Doli, Evdoxia–1927
Donahue, Carly M.–1754
Dong, David W.–1651, 1712
Dong, Qian–1863, 1864
Dongdong, Lu–1888
D'Onofrio, Kristen–1877
Donoso, Luis–1654
Doria, Damian–1697, Cochair Session 2aAAa (1695)
Dossey, Ellen–1930
Dosso, Stan E.–1801, 1886
Dottin-Haley, Brandon–1923
Dougherty, Kelsey–1663
Douglass, Alexander S.–1732
Dowd, Elliot–1902
Dowling, David R.–1706, 1732, 1792, 1887, 1936
Dowling, Zara–1775
Drinkwater, Bruce W.–1715
Driscoll, Michael–1864
D'Spain, Gerald L.–1793
Du, Hualong–1893
Du, Xuhao–1855
Dubbs, Lindsay–1856
Dubno, Judy R.–1718, 1821, 1822
Duda, Timothy F.–1886, Cochair Session 1aAO (1653)
Duffner, Pascal–1661
Dugan, Sarah–1794, 1799
Dugger, Edward–1658
Dumoulin, Charles L.–1891
Duncan, Joseph–1826
Dunn, Lincoln–1872
Dupont, Samuel–1896
Dupre, Aurelien–1862
Durach, Maxim–1903
Durofchalk, Nicholas C.–1771, 1935
Dutta, Indranil–1929
Dutta, Kelsey J.–1783
Dwyer, Robert–1796

Dzieciuch, Mathew–1803
Dzieciuch, Matthew A.–1803
Eary, Kathryn J.–1862
Eastland, Grant–1780
Eckersley, Casey–1738
Eckman, Emmet–1741
Eddins, Ann C.–1663
Eddins, David A.–1663, 1722, 1758, 1797, 1877
Edelmann, Geoffrey F.–1669
Edwards, Morgan–1731
Egan, Thomas–1674
Egnuni, Teklu–1810
Eichwald, John–1866
Elbing, Brian R.–1901
Elhilali, Mounya–1923
Elia Forte, Antonio–1907
Elkington, Peter–1769
Elliott, Andy S.–1792
Elliott, Lauren–1826
Ellis, Gregory M.–1876, Chair Session 2aPPb (1718)
Elsaadawy, Hossam–1733
Elston, Cody–1909
Emelianov, Stanislav–1779
Emmanouilidou, Dimitra–1923
Engelberg, Jonathan W.–1776
Erdman, John W.–1921
Erisman, Brad–1807
Escobedo, Julio A.–1901
Esfahlani, Hussein–1761
Etter, Paul C.–1804
Everbach, E. Carr–1778, 1811, Cochair Session 3aBA (1777), Cochair Session 3pBAb (1810)
Evers, Läslo G.–1868
Exner, Agata–1892
Fackler, Cameron J.–1805
Fagan, Mary K.–1731
Faircloth, Laura R.–1927
Fan, Huajun–1768
Fan, Langchen–1719
Fan, Xudong–1817
Fang, Jian–1864
Fang, Lezheng–1725
Fang, Nicholas X.–1792
Farber, Gregory S.–1866
Farbos de Luzan, Charles–1795
Farhadi, Afagh–1719
Fatemi, Mostafa–1891, 1893
Faytak, Matthew–1927
Fee, David–1868
Feehan, Colette–1930
Feenaughty, Lynda–1797
Feeny, Graham T.–1796
Feinberg, Jack–1750
Feist, Christopher–1806
Feng, Huihua–1882
Ferguson, Sarah H.–1915
Ferjan Ramirez, Naja–1765
Feth, Lawrence L.–1723
Feugère, Lionel–1920
Feuillade, Christopher–1654
Fiebig, André–1678, 1752
Filipov, Evgueni T.–1669
Finan, Donald–1816
Finette, Steven I.–1886
Finn, Anthony–1864, 1903
Finneran, James J.–1856
Finol, Ender A.–1746
Fischer, Jost L.–1709
Fischer, R. W.–Chair Session (1734)
Fishman, Glenn I.–1701
Fitzgerald, Laura–1709
Flamme, Gregory–1815, 1816
Flege, James E.–1826
Flego, Stefon M.–1930
Flesher, Brooke–1663
Flynn, Tyler J.–1706, 1792
Fogerty, Daniel–1718, 1822
Folkerts, Monica L.–1720
Foo, Wen-Chi–1859
Forrest, Jon–1933
Fox, Robert A.–1795, 1826
Frajuca, Carlos–1668
Francis, Alexander L.–1659
Franklin, Austin–1710
Franzoni, Linda P.–1889
Frazier, David L.–1723
Frazier, William G.–1661, 1867
Freear, Steven–1810
Freyman, Richard L.–1721, 1872
Friend, James–1787
Fristrup, K.–Vice Chair ASC S3/SC 1 Session (1772)
Fristrup, Kurt M.–1679, 1753
Froberg, James–1894
Fronk, Matthew–1725
Fruehwald, Josef–1931
Fu, Zhanao–1914
Fuchi, Kazuko–1668
Fuchs, Susanne–1931
Fullerton, Jeffrey–1652
Fulop, Sean A.–1930
Funk, Rob–1902
Furst, Miriam–1685
Fushimi, Tatsuki–1715
Gabriel, Jack D.–1677
Gabrielson, Thomas B.–1770, 1869
Gallagher, Hilary–1680
Gallot, Thomas–1726, 1903
Gallun, Frederick J.–1722, 1758, 1877
Gamba, Marco–1919
Gandhi, Pratik–1733
Gao, Hexuan–1823
Garrett, Steven L.–1682, 1924
Gaston, Jeremy–1722
Gaudette, Jason E.–1744
Gaudrain, Etienne–1720, 1821, 1914
Gautier, Francois–1726
Gay, Jennifer–1724
Ge, Chunsha–1732
Gebhardt, Jan–1679
Gee, Kent L.–1662, 1704, 1713, 1749, 1755, 1816, 1870, 1884, 1897, Cochair Session 2aPA (1713), Cochair Session 2pID (1749), Cochair Session 2pPA (1754)
Geib, Nate–1687
Gendin, Daniel–1891
Gendron, Paul J.–1802, Cochair Session 3aSP (1800)
Genstler, Curtis–1746
Genuit, Klaus–1679, 1752, 1898, 1917, Cochair Session 2pNS (1752), Cochair Session 5aAA (1917)
George, Parker–1811
Gerard, Nikhil J.–1761
Gerg, Isaac–1771
Gerhardson, Tyler–1862
Geroski, David J.–1936
Getz, Laura–1789
Ghaffarivardavagh, Reza–1726, 1896
Ghahrahmani Z., Elmira–1862
Ghai, Shweta–1731
Ghasemzadeh, Hamzeh–1690
Ghavami, Siavash–1891, 1893
Giacoma, Cristina–1919
Gibson, Gabriella–1920
Giegold, Carl–1739
Gifford, René–1877
Gifford, René H.–1796
Gilbert, Suzanna–1884
Gilkey, Robert H.–1722, 1874
Gillard, Jessica–1723
Gillman, Andrew–1668
Ginares, Alejandro–1903
Giordano, Nicholas–1677
Giovannone, Nikole–1796
Gladden, Josh R.–1706, 1880, 1924
Glowacki, Oskar–1887
Goates, Caleb B.–1791, 1884
Gockel, Hedwig E.–1721
Godefroy, Guillaume–1779
Godin, Oleg A.–1670, 1671, 1708, 1770, 1869
Godoy, Ricardo–1784
Goergen, Craig J.–1702, 1746
Gogoi, Pamir–1929
Golden, Matthew V.–1711
Goldrick, Matthew–1798
Goldsberry, Benjamin M.–1687
Goldsworthy, Michael–1744
Gonzalez, Valeria–1804
Good, Kenneth W.–1651, 1917, Cochair Session 1aAA (1651), Cochair Session 5aAA (1917)
Goodheart, Matthew–1784
Gordon-Salant, Sandra–1705, 1821
Goshorn, Edward L.–1797, 1878
Goshorn, Jennifer–1797, 1878
Goupell, Matthew–1705, 1759, 1876
Goupy, Alexandre–1900
Gouzoules, Harold–1776
Grace, Sheryl–1926
Graham, Alan–1756
Grant, Kenneth–1722
Grant, Ken W.–1721
Graves, Jackson–1784
Green, David B.–1905
Greene, Nate–1717
Greenhall, John–1756, 1925
Gregg, Joshua F.–1925
Gregory, Adriana–1893
Grigorieva, Natalie S.–1691
Grimm, Peter D.–1862
Grosh, Karl–1687, 1908, 1882
Grosz, Steven A.–1861
Gruensteidl, Clemens–1791
Gu, Juanjuan–1715
Guan, Steven–1811
Guers, Manton–1791
Guest, Daniel–1782
Guidi, Francesco–1892
Guild, Matthew D.–Cochair Session 4aEA (1863), Cochair Session 4pEA (1895)
Guilleminot, Johann–1792
Guilment, Thomas–1856
Guo, Chunli–1795
Guo, Jia–1699
Guo, Siyan–1797
Guo, Yueqi–1785
Guo, Yuting–1931
Gupta, Ankush–1926
Gutmark, Ephraim–1795
Guzzomi, Andrew–1855
Haberman, Michael R.–1687, 1707, 1726, 1727, 1754, 1812, 1924, Cochair Session 2pID (1749), Cochair Session 4aEA (1863), Cochair Session 4pEA (1895)
Hadley, Judith–1892
Hagedorn, Anna–1663
Hahn, Allison H.–1807
Haile, Eyobel–1902
Hake, Alison–1882
Haley, Rebecca–1928
Hall, Hubert S.–1824, Cochair Session 3pSA (1823)
Hall, Neal A.–1707
Hall, Timothy J.–1891
Hall, Timothy L.–1862, 1925
Hambric, Stephen–1794
Hamery, Pascal–1661
Hamilton, Mark F.–1707, 1713, 1715, 1754, 1755, 1756, 1811, 1924
Hammans, Christine A.–1730, 1764
Hammond, Ross–1853
Han, Aiguo–1921
Han, Yuting–1801
Hanford, Amanda–1725
Hanson, Elizabeth–1777
Harkin, Sean–1711
Harne, Ryan L.–1669, 1714, 1725, 1823, Cochair Session 1aSP (1668)
Haro, Stephanie–1744
Harp, Andrew G.–1869
Harris, Natalie–1881
Hart, Carl R.–1661, 1800, 1885
Hartmann, William M.–1813, 1872
Hasegawa, Yoshimi–1657
Hasenfratz, Liat–1730
Hashemi Hosseinabad, Hedieh–1933
Hasson, Uri–1730
Hawes, David–1792
Hawkes, Menley R.–1901
Haworth, Kevin J.–1703, 1704, 1705, 1746, 1747, Chair Session 3pBAa (1808), Cochair Session 2aBA (1701), Cochair Session 2pBA (1745)
Hayward, Chris–1660
He, Xiao–1768
He, Yuan–1721, 1871
He, Yunjuan–1825
Heald, Shannon L.–1688
Healy, Eric–1874
Hebb, Megan–1876
Heffner, Christopher C.–1790
Hegarty, Gerrick–1751
Heinz, Michael G.–1663, 1716, 1717
Helder-Hoek, Lean–1774
Helfer, Karen S.–1872
Hellweg, R. D.–Chair Session (1734)
Helmholz, Hannes–1889
Hendley, Samuel A.–1746, 1747

Hengy, Sébastien–1661
Henri, Fabiola–1928
Henry, Kenneth S.–1719, 1879
Hernández, Álvaro–1884
Hernandez, Christopher–1892
Herstein, Peter–1807
Hess, Christi–1906
Hetzer, Claus–1867
Hibben, Rebecca–1763
Hickey, Craig–1863
Hickey, Michael P.–1901
Higgins, Alex–1781
Hildebrand, John–1919, Chair Session 5aAB (1919)
Hildebrandt, Tamarae–1911
Hill, Adam J.–1853
Hill, Thomas L.–1715
Hitchcock, Elaine R.–1799
Hjortnaes, Nils–1928
Hoeft, Michael–1805
Hoffman, Elissa–1874
Hoffmann, Maureen–1934
Hoffmeister, Brent K.–1893
Hofmann, Alex–1677
Hoglund, Evelyn M.–1723, Cochair Session 3pPPa (1819)
Holland, Charles W.–1801, 1887
Holland, Christy K.–1705, 1745, 1746, 1747
Hollmann, Joseph L–1810
Holm, Sverre–1771, 1861, 1884
Holmberg, Derek E.–1923
Holt, Alexis–1878
Holt, Lori L.–1766
Holt, R. Glynn–1810, 1860, 1926
Holt, Rachael F.–1798
Holt, Yolanda F.–1819
Holtz, Allison–1722, 1874
Hong-lun Yeung, Alex–1826
Hoole, Philip–1914
Hoover, Eric C.–1705, 1722, 1758, 1877
Hoover, K. Anthony–1696, Cochair Session 2aAAa (1695)
Horn, Timothy–1893
Hossain, Md.–1664
Hots, Jan–1793
Hou, Zhengyu–1671
Houle, Nichole–1934
Houser, D. S.–Vice Chair ASC S3/SC 1 Session (1772)
Houston, Derek–1729, 1766
Houte, Kayla M.–1733
Howard, Daniel R.–1706
Howarth, Thomas R.–1863
Hoyt, Travis N.–1712
Hsiao, Ying–1876
Hsu, Timothy–1855
Hu, Fang Q.–1817
Huang, Lixi–1867
Huang, Tony Jun–1666, 1786
Huang, Wanting–1879
Huang, Yu–1898
Huangpu, Li–1936
Huet, Moïra-Phoebé–1720
Huff, Emma–1786
Huffman, Arthur–1682
Hughes, Michael–1811
Hui, Juan–1693, 1732, 1936
Hull, Andrew J.–1792
Hull, Kirsten–1739, 1740, Cochair Session 2pAA (1738)
Hummel, Kenton–1658
Humphreys, Luke–1896
Hunt, Samuel–1751
Husain, Fatima–1716
Husain, Fatima T.–1907
Huyck, Julia J.–1724
Hwang, Jiwon–1826
Hwang, Young–1928
Iaboni, Andrea–1658
Iezzi, Alexandra M.–1868
Iglesias, Digno–1733
Ihlefeld, Antje–1759, Cochair Session 4pPP (1904)
Ilinskii, Yurii A.–1715
Inchingolo, Riccardo–1656
Ingram, Nicola–1810
Ionascu, Dan–1891
Irarrazabal, Francisco J.–1870
Isakson, Marcia J.–1707
Isarangura, Sittiprapa–1722
Iskander-Rizk, Sophinese–1778
Iyer, Nandini–1879
Jacewicz, Ewa–1795, 1826
Jacoby, Nori–1784
Jaggers, Zachary–1912
Jahanpanah, Newsha–1703, 1704
Jahn, Kelly N.–1905
Jallah, Johnetta–1864
James, Deryck D.–1662
Jamsek, Izabela A.–1798
Janis, Brett R.–1895
Jaramillo, Ana M.–1853, Cochair Session 4aAA (1853)
Jasinski, Christopher–1792
Jech, J. Michael–1653
Jenkins, Kimberly–1722
Jenkins, Scott A.–1793
Jennings, Skyler G.–1717, 1723
Jennings, Todd R.–1876
Jeon, Miyoung–1724
Jeremic, Aleksandar–1894
Jerome, Thomas S.–1715
Jerome, Trevor W.–1794, 1815
Ji, Jun–1726
Ji, Yunjia–1768
Jiang, Bing'er–1913
Jiang, Changyong–1867
Jiang, Jingfeng–1675
Jiang, Weikang–1898
Jin, Yabin–1762
Jing, Yun–1715, 1726, 1761, 1762
Jiu, Pennine–1699
Joasil, Arielle–1733
Johnsen, Eric–1925
Johnson, Cynthia–1763
Johnson, Eric–1874
Johnson, Gregory–1931
Johnson, Jay R.–1672
Johnson, Michael T.–1931, 1932
Johnson, Pablo–1733
Johnson, Paul A.–1754
Johnson, Paul T.–1924
Johnson, Shawn–1771
Johnstone, David A.–1891
Jones, Adrian D.–1770
Jones, L. Keeta–1865, Cochair Session 3aED (1782), Cochair Session 3eED (1830)
Jongman, Allard–1767, 1828, 1911, 1915
Juhl, Abigail T.–1686
Kabakoff, Heather M.–1796, 1827
Kahn, David–1738, 1740
Kalashnikova, Marina–1728
Kalra, Sanjay–1673
Kamal, Sarah–1902
Kamrath, Matthew–1661
Kanter, Shane J.–Cochair Session 3pAA (1805)
Kao, Chieh–1764
Kapadia, Alexandra M.–1915
Karayanidis, Frini–1767
Karbalaeisadegh, Yasamin–1860, 1861
Kardous, Chucri A.–1815
Kargl, Steven G.–Chair Session 1pUW (1691)
Karunarathne, Ashoka–1924
Kastelein, Ronald–1774
Kates Varghese, Hilary–1671
Kathiresan, Thayabaran–1766
Katsnelson, Boris–1670, 1671, 1770
Katz, William F.–1927
Kawasaki, Jonathan–1806
Kazemi, Arash–1675
Ke, Guoyi–1781
Keerstock, Sandie–1911
Keil, Nathan–1784
Kelley, Matthew C.–1911, 1934
Kelly, James F.–1861
Kennedy, Jermaine L.–1691, 1864
Keolian, Robert M.–1682
Kessler, David–1877
Ketterling, Jeffrey A.–1701
Khan, Amir–1811
Khan, Rafay–1716
Kharlamenko, Julia–1827
Khokhlova, Vera A.–1714, 1857, 1902
Khosla, Sid M.–1795
Khyam, Mohammad Omar–1742, 1743
Kidd, Gary R.–1795, 1909
Kidd, Gerald–1719, 1737, 1876
Kidner, Brooke L.–1927
Kiefer, Daniel A.–1791
Kim, Bong Young–1896
Kim, Heejin–1795
Kim, Jiseung–1933
Kim, Kang–1703, 1858, 1859, Cochair Session 4aBAa (1857), Cochair Session 4pBAa (1891)
Kim, Kimteri–1652
Kim, Michael–1824
Kim, Minji–1786
Kim, Myungsook–1922
Kim, Subong–1717
King, Eoin A.–1733, 1884
Kingston, John–1913
Kinney, Kaelin–1765
Kirby, Jessica–1763
Kirsteins, Ivars P.–1692, 1801
Kist, Matthew–1799
Klang, Alix–1875
Klein, McKenzie–1906
Kleinschmidt, Dave F.–1689, 1910
Kline, Brian L.–1794
Klock, John–1857
Klockgether, Stefan–1720
Kluk, Karolina–1907
Knight, Derrick P.–1899
Knobles, David P.–1801, 1935, 1936
Knutsen, Sten–1910
Kobrina, Anastasiya–1705, 1718, 1806, 1807
Koch, Robert M.–Cochair Session 3aSAb (1793), Cochair Session 4aSA (1880)
Koenig, Laura L.–1799, 1931
Kolacz, Jacek–1764
Kolios, Michael C.–1892
Kondaurova, Maria V.–1705, 1731, 1765
Kontopidis, Patrick–1709
Koo, Chi Wan–1673
Kopechek, Jonathan A.–1746, 1894, 1895, Cochair Session 2aBA (1701), Cochair Session 2pBA (1745)
Korman, Murray S.–1681, 1918
Korostenski, Larissa–1767
Kosarev, Victor–1769
Krasnow, Max M.–1766
Kreft, Heather A.–1875
Kreider, Wayne–1857
Kreiman, Jody E.–1930
Kremin, Lena V.–1766
Krishfield, Richard A.–1803
Krishnamurthy, Siddhartha–1899
Kruizinga, Pieter–1778
Krull, Vidya–1876
Kubli, Lina R. R.–1721
Kuc, Roman B.–1741, 1774
Kuhl, Patricia K.–1765
Kumar, Ajith U.–1906
Kumar, Krishna–1703
Kumar, Krishna N.–1787, 1817
Kumar, Raj–1762
Kumar, Viksit–1893
Kumareswaran, Mitra–1731
Kummer, Ann W.–1933
Kuo, Jennifer–1927
Kurdila, Andrew–1699
Kwan, James J.–1745, 1892, 1894
Kwan, Timothy–1664
Kwon, Harim–1931
Kwon, Hyung-Suk–1687, 1761
Kye, Joong Eup–1935
Labadie, Robert–1796
Labuda, Cecille–1706
Lachowski, Kacper–1779
Lafond, Maxime–1745
Lai, Puxiang–1778
Lally, Michael F.–1795
Lam, Po–1859
Lamba, Michael A.–1891
Lamont, Andrew–1913
Landis, Steven J.–1854
Landschoot, Christopher–1802
Lane, Alison–1767
Lane, Shelly–1767
Langhirt, Mark–1887
Langley, Robin S.–1792
Lani, Shane–1896
Lankford, James E.–1816
Lanning, Joshua M.–1822
Lantz, Brian–1667
Larin, Kirill–1675
Larson, A. Noelle–1893

Lauer, Amanda–1718, Cochair Session 4pPP (1904)
Lavandier, Mathieu–1722, 1872
Lavery, Andone C.–1653, 1706
Lavery, Richard–1883
Lawless, Martin S.–1883, 1900
Leahy, Stephane–1698
Leamy, Michael J.–1725
Le Bas, Pierre-Yves–1755
LeBlanc, Camille–1733
Le Pichon, Alexis–1868
Lee-Rubin, Harry–1766
Lee, Abe H.–1670
Lee, Adrian K. C.–1758
Lee, Brandon M.–1887
Lee, Dong Hyun–1709
Lee, Hyeon–1737, 1743, 1774
Lee, Jayden–1912
Lee, Joy–1826
Lee, Jungmee–1683, 1871
Lee, Kevin M.–1672, 1924, Cochair Session 1pAO (1670), Cochair Session 5aPA (1923)
Lee, Michael–1727
Lee, MingYang–1883
Lee, Seungjun–1883
Lee, SungTae–1864, 1865
Lee, Yoonjeong–1930
Leek, Marjorie R.–1821
Lehet, Matt–1766
Lei, Wang–1888
Leibold, Lori–1790, 1879, 1904
Leishman, Timothy W.–1704, 1889
Lelo de Larrea Mancera, Esteban S.–1877
Lembke, Sven-Amin–1785
Lemons, Charlsie–1719
Lenihan, Kiegan–1882
Lenox, Mark–1857
Lentz, Jennifer–1683, 1721, Cochair Session 1pPP (1683)
Lenzi, Arcanjo–1881
Leong-Poi, Howard–1745
Lesser, Thomas G.–1656
Leventis, Nicholas–1824
Levi, Susannah V.–1690, 1827, 1909, 1910, 1933, 1934, Chair Session 3pSC (1825)
Levy, Erika–1827
Lewandowski, Marcin–1701
Lewis, Gregory F.–1764
Lew-Williams, Casey–1730, 1765
LI, Bei–1825, 1929
Li, David S.–1779
Li, Gang–1667
Li, Huangpu–1732
Li, Hui–1659, 1805
Li, JiangQiao–1732, 1936
Li, Jin–1693
LI, Jingtao–1795
Li, Jinyu–1692
Li, Junfei–1666, 1687, 1762, 1882
Li, Kun–1855, 1856
Li, Mingshuang–1828, 1934
Li, Qi–1692, 1886
Li, Qian–1922
Li, Sarah R.–1794, 1799
Li, Xiaodong–1817
Li, Xiukun–1693
Li, Yang–1795
Liang, Laura–1798
Ligon, David–1662
Lilley, Kevin D.–1910
Lim, Sung-Joo–1822, 1915
Lin, Iris–1915
Lin, Kevin Y.–1880
Lin, Ying-Tsong–1671, 1803, 1886, 1887
Lind, S. J.–Vice Chair, ASC S12 Session (1773)
Linke, Simon–1676
Linnett, Laurie–1769
Lipani, Lisa–1932
Lipkens, Bart–1787, 1817
Lirette, Robert L.–1818, 1863
Litovsky, Ruth–1750, 1873, 1906
Liu, Chang–1828, 1934
Liu, Haijun–1863, 1864
Liu, Jinpeng–1692
Liu, Jizhou–1817
Liu, Ting-Wei–1670, 1687
Liu, Yanming–1699
Liu, Yue–1818
Liu, Yunbo–1922
Llanos, Fernando–1820
Lockhart, Stephen B.–1856
Lofqvist, Anders–Cochair Session 2pID (1749)
Long, Gayle–1795
Long, Glenis R.–1757
Lonzaga, Joel B.–1714
Loomba, Rohit–1921
Lopez, Christian A.–1901
Loranger, Scott–1653
Lordan, Marius Catalin–1730
Loshin, Ben–1874
Lott, Francois–1869
Lotven, Samson A.–1928
Loubeau, Alexandra–1899
Loverde, John–1651, 1712
Lowenstein, Joanna H.–1822
Lu, Hongbing–1824
Lu, Jie–1763
Lu, Ning–1862
Lu, Shouwei–1882
Lu, Zhiqu–1880
Luchies, Adam–1702
Lucor, Didier–1900
Lulich, Steven M.–1795, 1928, 1930
Luna-Acosta, Andrea–1804
Lundt, Jonathan E.–1862
Lunkov, Andrey–1671
Lunsford, Sierra–1741
Lutfi, Robert–1683, 1871
L'vov, M.– Session (1734)
Lyons, Anthony P.–1653, 1706, 1770
Lyons, Gregory W.–1661, 1870, Cochair Session 1aPA (1660), Cochair Session 1pPAa (1680)
Lysenko, Eric J.–1901, 1904
Lytle, Sarah–1765
Ma, Chu–1792
Ma, Lin–1732
Macelloni, Leonardo–1655
Macpherson, Ewan A.–1724, 1880
MacWhinney, Brian–1729
Maddox, Alexandra–1795
Maddox, Ross K.–1759, 1908
Madigosky, W.–Chair Session (1734)Madison, Jeffrey–1712
Magalhaes, Nadja S.–1668
Magliula, Elizabeth A.–Cochair Session 3aSAb (1793)
Mahalingam, Neeraja–1794
Maher, Sean–1869, 1901
Mai, Alexandra–1663
Main, Evan N.–1893
Majdi, Joseph–1921
Malakooti, Sadeq–1824
Malik, Bilal–1857
Mallik, Sanku–1894
Malone, Matthew–1864
Malphurs, David E.–1864
Malykhin, Andrey–1770
Mamou, Jonathan–1859, Cochair Session 4pBAb (1892)
Mamun, Nursadul–1664
Mancia, Lauren–1925
Manik, Henry M.–1691
Manocha, Dinesh–1890
Mansour, Naim–1873
Mapp, Peter–1853
Marchitelli, Stefania–1933
Marcillo, Omar–1901
Markley, Kely J.–1732
Marks, Kendra L.–1873
Marques, Luciana–1926
Marques, Manuel–1811
Marr, Jeff–1806
Marschall, Márton–1873
Marsili, Enrico–1894
Marston, Philip L.–1692
Martin, Charlie–1741
Martinelli, Sheri–1887
Maruvada, Subha–1922, Cochair Session 3aCA (1780)
Marzo, Asier–1715
Mason, Christine–1719, 1737, 1876
Mast, T. Douglas–1705, 1794, 1799, 1812, 1862, 1891, Cochair Session 4aBAa (1857), Cochair Session 4pBAa (1891)
Masterson, Jack A.–1794, 1799
Matlack, Kathryn H.–1666, 1760, Cochair Session 1aSAa (1665), Cochair Session 1pSA (1686)
Matoza, Robin S.–1869, 1901
Matriss, Anthony–1733
Matthews, Cameron A.–1775
Mattingly, Michelle M.–1905
Matula, Thomas–1779
Maurer, Alex–1652
Maxwell, Adam D.–1746, 1747
May, Tobias–1873
Mayell, Marcus R.–1740
Mayer, Larry–1653, 1671
Maynard, Julian D.–Chair Session 1pPAb (1682)
Mayo, Nathanael K.–1863
McAllister, Tara–1796
McAuley, J. Devin–1909
McAuliffe, Michael E–1931
McCarthy, Ryan–1884
McCluskey, Connor–1791
McComas, Sarah–1660, 1900
McCormick, Cameron A.–1823
McCreery, Ryan W.–1879, 1904
McDermott, Josh–1721, 1784
McDonough, Grace–1926
McFayden, Tyler–1764
McGee, JoAnn–1806
McGough, Robert–1860, 1861, Chair Session 1aBAa (1655)
McGough, Robert J.–Cochair Session 4aBAb (1860)
McGowan, Kevin B.–1910, 1931, Chair Session 4pSC (1908)
McKay, John D.–1935
McKay, Margaret G.–1869, 1901
McKenna, Mihan H.–1660, 1870
McLane, Daniel–1923
McLane, Ian–1923
McLane, Jaishree–1923
McLaughlan, James R.–1810
McLaughlin, Drew J.–1827
McLaughlin, Michaela–1797
McMurray, Bob–1730, 1764
McNeese, Andrew R.–1672
McNeill, Joshua–1932
McPherson, Malinda J.–1721, 1784
McQueen, James M.–1915
McQuown, M. S.–1769
Meacham, J. Mark–1786
Meade, Joshua–1903
Meaud, Julien–1665, 1719
Medda, Alessio–1902
Meggitt, Joshua–1792
Mehr, Samuel A.–1766
Mehta, Anahita H.–1664, Cochair Session 1aPP (1663), Cochair Session 2aPPa (1716), Cochair Session 3pPPa (1819)
Meinke, Deanna K.–1816
Meir, Avishai–1924
Melodelima, David–1862
Melon, Manuel–1896
Menard, Lucie–1928
Menze, Michael A.–1895
Mercado-Shekhar, Karla P.–1703
Mercado, Eduardo–1807, 1878
Meves, Alexander–1676
Meyer, Richard J.–1771
Michalis, Krista–1889
Michalopoulou, Zoi-Heleni–1801, Cochair Session 3aSP (1800)
Micheyl, Christophe–1720
Middlebrooks, John C.–1684
Mielke, Jeff–1931
Mielnicki, Nicole–1878
Mikhalvesky, Peter–1804
Miksis-Olds, Jennifer L.–1654, 1671, 1706
Milbrandt, Todd A.–1893
Milekhina, Olga–1719
Miles, Ronald–1698
Milkey, Casey D.–1721
Miller, Andrew N.–Cochair Session 2aAAb (1697)
Miller, Gregory A.–1740
Miller, James D.–1795, 1821
Miller, James H.–1671, 1672, 1707
Miller, Jennifer R.–1695, 1738, 1739, 1918
Miller, Max–1824
Miller, Rachel E.–1718, 1822
Millet, Christophe–1869, 1900
Milliren, Christopher–1806
Ming, Chen–1737, 1743, 1744
Misurelli, Sara–1906
Mittal, Vijay–1798
Mix, Doran–1858
Mobadersany, Nima–1892

Moberly, Aaron C.–1876
Mobley, Frank S.–1680
Mobley, Joel–1706, 1818, 1863, 1880
Modarres-Sadeghi, Yahya–1775
Mohammadi, Mohammad–1863
Mohanty, Kaustav–1674, 1860
Monahan, Philip J.–1914
Mondain-Monval, Olivier–1762
Monson, Brian B.–1723, 1767, 1874
Monto, Nicholas R.–1914
Mookherjee, Mainak–1925
Moon, Lori A.–1795
Moore, Andrew–1920
Moore, Ashley–1790
Moore, Brian C.–1783, 1897
Moore, Jesse T.–1671
Moore, Thomas–1677, 1708, 1709
Moore, Travis M.–1871
Moorhouse, Andy–1792
Morales, Nicolas–1890
Moran, Steven–1676
Moreira, Jennifer M.–1926
Mores, Robert–1676
Morgan, Mallory M.–1737
Morgan, Shae D.–1705, 1912
Moriarty, Vincent–1737
Morrill, Tuuli–1789
Morris, D. Cody–1859
Morris, Hayley–1878
Morris, Zakayo Ndiku–1801
Morrison, Evan L.–1856
Mortenson, Michael C.–1884
Moser, Cody J.–1766
Muehleisen, Ralph T.–1886
Muglia, Mike–1856
Muhlestein, Michael B.–1726
Muir, Thomas G.–1755
Mukai, Yoichi–1912
Muller, Marie M.–1674, 1781, 1860, 1861, 1893
Müller, Rolf–1698, 1699, 1700, 1741, 1742, 1743, 1744, 1774, Chair Session 2aAB (1698), Chair Session 2pAB (1741), Chair Session 3aABa (1774)
Münster, Malte–1709
Mulsow, Jason–1856
Mun, Lucas–1742
Munson, Benjamin–1789, 1798
Murphy, Emily M.–1895
Murphy, W. J.–Chair ASC S3 Session (1772)
Murphy, William J.–1662, 1815, 1816, 1866, Cochair Session 3pNS (1815), Cochair Session 4aNS (1865)
Murray, Todd W.–1791
Myers, Emily B.–1790
Myers, Jennifer–1722
Myers, Joseph W.–1739, 1740
Myers, Kyle R.–1880
Myers, Samantha–1928
Naderyan, Vahid–1863
Naify, Christina J.–1727, Chair Session 3pID (1812), Cochair Session 2aSA (1725), Cochair Session 2pSA (1760)
Naik, Aneesh–1913
Najjuuko, Samusha–1902
Natesan, Rajni–1857
Nayak, Rohit–1891
Nazeri, Arash–1862
Neale, William–1890
Nechaev, Dmitry–1719, 1776
Neilsen, Traci–1889, 1904
Neilsen, Tracianne B.–1704, 1708, 1869, 1884, 1901, 1935, 1936
Neldner, Lauren–1709
Nelson, J. T.–Chair ASC S2 Session (1694)
Nelson, Pauline–1870
Nelson, Peggy B.–1806, 1879
Nemati, Navid–1792
Nenadic, Filip–1911
Nencheva, Mira L.–1765
Neu, Corey P.–1702
Newhall, Arthur–1671, 1886
Ngnie Kamga, Irvane–1811
Nguyen, Sarah–1920
Nguyen, Trong–1861
Nichols, Matt–1739
Niedert, Elizabeth E.–1702, 1746
Niehl, Annika–1659
Nightingale, Kathryn–1859
Nikolaev, Dmitry A.–1857
Nikolajczyk, Jacob–1726
Nisha, K. V.–1906
Nittrouer, Susan–1822
Noble, Jack H.–1796
Noble, John–1662
Noble-Vranish, Misty L.–1746
Nolan, Lindsay–1765
Norceide, Flore–1817
Norris, Andrew–1727
Nothaft, Julia–1759
Novakovich, Daniel J.–1870
Nusbaum, Howard C.–1688
Nygaard, Lynne C.–1689
Oakley, Madeleine–1826
Oberai, Assad–1891
Obi, Chester–1733
O'Brien, Gabrielle–1760
O'Brien, W. D. Jr., –1921
O'Donnell, Matthew–1779
O'Donnell, Tim–1913
O'Driscoll, Martin–1907
Oeding, Kristi–1879
Oelze, Michael–1715, 1859, 1921
Oelze, Michael L.–1861, 1883
Offner, Avshalom–1924
Ogawa, Takuya–1859
Oh, Jasper–1876
Oh, Miran–1926
Ohm, Won-Suk–1755, Cochair Session 2aPA (1713), Cochair Session 2pPA (1754)
Olds, Daniel–1729, 1764
Olgun Kucuk, Hilal–1673
Oliver, Jordan N.–1659
Olsen, Rachel M.–1930
Olson, Bruce C.–Cochair Session 4aAA (1853)
Olson, Bruce–1853
Olson, Derek R.–1770, Chair Session 2pUW (1770)
Olson, Stephen T.–1794
Onwubiko, Stephen G.–1708
Oren, Liran–1795, 1796, 1933
Orena, Adrial J.–1765
Orena, Adriel John–1766
Ortega, Nicholas D.–1899
Osborn, Jenna–1894
Osborn, Thomas–1673
Oschkinat, Miriam–1914
Oshima, Atsushi–1768
Osmanski, Michael–1785
Ossandon, Tomas–1784
Ostashev, Vladimir–1900
Ostashev, Vladimir E.–1661, 1800
Ostrovskii, Igor–1755
Ostrovsky, Ilia–1671
Ott, Olivia–1896
Oxenham, Andrew J.–1664, 1719, 1782, 1784, 1788
Ozmeral, Erol J.–1663
Padilla, Alexandra M.–1653, 1692
Paek, Hyun–1695, 1738, 1739, 1918
Palandrani, Katherine–1663, 1722
Palcich, Hunter A.–1703
Palmeri, Mark L.–1859
Pàmies-Vilà, Montserrat–1677
Pan, Jie–1855
Pan, Xiang–1802
Pandey, Aditya–1862
Pandompatam, Govind–1673
Pankonien, Alexander–1668
Panneton, Robin–1764
Pantea, Cristian–1756, 1925
Paoletti, Dennis–1695
Paratore, Trevor–1924
Pardo, Jennifer–1910
Parida, Satyabrata–1716
Parizet, Etienne–1720
Park, Han Jung–1924
Park, Hyung Woo–1864, 1865
Park, Joonho D.–1935
Park, Kwanho–1864, 1865
Park, MinSeuk–1936
Park, Seongjin–1827, 1915
Park, Seung-Soo–1935
Park, Taeyoung–1755
Park, Yongsung–1936
Parshall, Alexandra M.–1776
Partain, Mark–1909
Paruchuri, Sai T.–1824
Pastore, M. Torben–1685, 1723, 1873, Cochair Session 3aMU (1782), Cochair Session 3pMU (1813)
Patankar, Harshal Prakash–1885
Patchett, Brian D.–1715
Patel, Raj–1738
Patil, Ganesh U.–1666, 1760
Patnaik, Sourav–1746
Patro, Chhayakanta–1875
Patterson, Roy D.–1783
Paulus, Yannis M.–1810
Peacock, John–1717
Pearson, Dylan V.–1910
Pecknold, Sean–1803
Pedersen, Katrina–1816
Pelat, Adrien–1726
Pele, Arnaud–1667
Pelivanov, Ivan–1779
Pellegrino, Elisa–1929
Penelet, Guillaume–1924
Peng, Xiuyuan–1882
Peng, Z. Ellen–1873, Cochair Session 4aPP (1871)
Penhale, Miles B.–1883
Peppin, R. J.–Chair ASC S1 Session (1736), Vice Chair ASC S2 Session (1694)
Perazio, Christina E.–1804
Pereira, Rebeca–1861
Perelman, Brandon–1722
Perez, Daniel–1792
Perkins, Russell–1817
Perol, David–1862
Perrachione, Tyler K.–1690, 1822, 1912, 1915, Cochair Session 1pSC (1688)
Perrier, Pascal–1928
Perrone, Michael–1784
Perry, Trevor T.–1879
Pescador Sarda, Alexandre A.–1881
Pestka II, Kenneth A.–1923
Petrin, Christopher–1901
Pettit, Chris L.–1800, 1885
Pettit, Erin C.–1887
Phillips, James E.–Cochair Session 1aSAb (1667), Cochair Session 2aNS (1710)
Phoon, Colin K.–1701
Piacsek, Andrew A.–1881
Piazza, Elise A.–1730, 1765
Pickens, Anne T.–1708
Pierce, Allan D.–1787, 1800
Pineda Mora, Gina M.–1827
Piol, Serena–1909
Pirshafiey, Nasser–1857
Pitt, Jonathan–1692
Pluymers, Bert–1888
Podkowa, Anthony–1859
Podoleanu, Adrian–1811
Polascik, Thomas–1859
Polka, Linda–1765, 1766, Cochair Session 2aSCa (1728)
Pompei, Joseph–1714
Poncelet, Olivier–1762
Ponder, Julia–1806
Ponschab, Michael–1791
Ponsot, Emmanuel–1663
Popa, Bogdan Ioan–1687, 1761, Cochair Session 1aSAa (1665), Cochair Session 1pSA (1686)
Popov, Vladimir–1776
Popovics, John–1756
Porges, Stephen W.–1764
Porretta, Vincent–1912
Potty, Gopu R.–1671, 1672, 1707, Cochair Session 1pAO (1670)
Poudel, Joemini–1858
Pozzo, Lilo–1779
Prather, Wayne E.–1661, 1880
Prendergast, Brendan–1909
Preston, Jonathan L.–1796
Priddy, Mariah C.–1746, 1894, 1895
Priyadarshan, Gautam–1880, 1924
Proshutinsky, Andrey Y.–1803
Prud'homme, Luna–1722, 1872
Pryor, Nina–1879
Puckett, Brian J.–1753
Pullan, Jessica–1894
Putman, Melanie–1664
Qiao, Wenxiao–1769
Qu, Allen–1822
Qu, Lei–1867
Quan, Li–1761
Quinn, Brian G.–1870

Rad, Parsa–1655
Radicchi, Antonella–1753
Rahman, Abdullah F.–1672
Rakerd, Brad–1872
Raley, Michael–1711
Raley, Sean–1781
Ramirez, Juan–1669
Ramon, Guy Z.–1924
Ramsey, Gordon P.–1709
Ramsay, Gordon–1731
Randrianarison, Rose Marie–1919
Ransom, Serena–1724
Rasband, Reese D.–1662
Raspet, Richard–1863
Rathsam, Jonathan–1899
Raudales, David–1889
Raymond, Jason L.–1778, 1811
Redmon, Charles–1915
Reed, Jessa–1731
Reed, Paul E.–1819
Reetzke, Rachel–1820
Rehmann, Julia–1720
Reichenbach, Tobias–1907
Reichman, Brent O.–1755
Reiff, Christian G.–1662
Reinisch, Eva–1914
Reit, Eric–1666
Remillard, Grace–1864
Remillieux, Marcel–1755
Renwick, Margaret E.–1932
Reuter, Eric L.–Cochair Session 1aAA (1651)
Reyes Herrera, Edwin D.–1825
Reynolds, D. D.– Session (1734)Rice, Scott–1894
Richards, Michael S.–1858
Richie, Sarah–1873
Ricketts, T.–Vice Chair ASC S3 Session (1772)
Riley, Michael A.–1794, 1799
Rimington, Dennis–1793
Rivoire, Michel–1862
Rizzi, Stephen A.–1899
Roa, Marylin–1695, 1738, 1739, 1918
Roan, Michael J.–1737, 1743, 1774, Chair Session 4aSP (1883)
Roberts, Brian–1872, 1909
Rodgers, Brooke–1708
Rodriguez, Brianna–1683, 1871
Rodriguez, Erik A.–1894
Rodriguez, Mauro–1925
Roesner, Imme–1909
Rogers, Jeffrey S.–1669, Cochair Session 1aSP (1668)
Rogers, Kevin–1903
Rohrbach, Daniel–1859
Rokni, Eric–1922
Roland, William–1696
Rollins, Michael–1796, Cochair Session 2aID (1704)
Romanchek, Brittany A.–1776
Romigh, Griffin D.–1879
Rosemberg, Celia–1763
Rosen, David–1675
Rosen, Merri–1724, 1905
Rosnitskiy, Pavel B.–1902
Ross-Johnsrud, Benjamin P.–1817
Ross, John P.–1910
Rouse, Jerry W.–1889
Roux, Olivier–1920
Roy, Ronald A.–Cochair Session 3aBA (1777), Cochair Session 3pBAb (1810)
Roy, Ronald–1778, 1811
Rozynova, Aleksandra–1888
Ruan, Yufang–1765
Rupitsch, Stefan J.–1791
Rush, Brittney M.–1859
Rusk, Zane T.–1900
Russell, Daniel A.–1707, Cochair Session 2aID (1704), Cochair Session 2pED (1748), Cochair Session 3aED (1782), Cochair Session 3eED (1830), Cochair Session 3pNS (1815)
Ryan, Emily–1926
Ryan, Teresa J.–1824
Rychtarikova, Monika–1888
Ryherd, Erica E.–1657, 1658, 1706, 1753
Rysling, Amanda–1913
Saarelma, Jukka–1889
Sabatini, Roberto–1901
Sabra, Karim G.–1654, 1771, 1935, 1937
Saccocia, Gwen O.–1878
Saenz, Eduardo–1768
Sagers, Jason D.–1672, 1803
Sakiyama, Naoki–1768
Saksena, Anpeksh A.–1714
Salloom, William–1718, 1878
Saloio, Jack–1787, 1817
Samir, Anthony–1922
Sances, John M.–1910
Sanderson, Richard–1869
Sangha, Gurneet S.–1702
Sankaran, Narayan–1813
Sapozhnikov, Oleg A.–1714, 1857, 1902
Sari, Jim–1896
Sarkar, Kausik–1703, 1892, 1894
Sartori, Daniel–1733
Sasmal, Aritra–1687, 1908
Satish, Aprameya–1937
Saunders, Melvin–1696
Savino, Anthony–1806, 1888
Savioja, Lauri–Cochair Session 4pAA (1888)
Sayeed, Ollie–1914
Sayles, Mark–1906
Scalzi, Ken–1686
Scarborough, David E.–1896
Scarborough, Rebecca–1926
Schade, Lars–1679
Scharine, A. A.–Vice Chair ASC S1 Session (1736)
Schaub, Mary–1765
Scheifele, Peter M.–1776, 1777, 1799, 1807
Schenker, Molly F.–1913
Scherer, Ronald–1929
Schilb, Emily–1658
Schlauch, Robert S.–1790
Schlunk, Siegfried–1702
Schmidt, Anna M.–1913
Schmidt, Sara–1716
Schnitta, Bonnie–1711, Cochair Session 2aNS (1710)
Schoen, Scott J.–1859
Schoentgen, Jean–1909
Schomer, P. D.–Vice Chair Session (1734)
Schrode, Katrina M.–1718
Schuette, Dawn–1740
Schulte-Fortkamp, Brigitte–1678, 1752, 1865, Cochair Session 1pNS (1678), Cochair Session 2pNS (1752), Cochair Session 4aNS (1865), Cochair Session 5aAA (1917)
Schulz, Anneliese K.–1874
Schumacher, Christian–1863
Schutz, Michael–1723
Schwartz, Jay W.–1776
Scinto, Jennifer–1711
Scott, Adrienne–1702
Scott, E. K. Ellington–1780
Scott, Hannah–1930
Screven, Laurel A.–1718
Seger, Kerri–1804
Seidl, Amanda–1690
Seitz, Aaron–1722, 1877
Semperlotti, Fabio–1670, 1687
Sen Gupta, Ananya–1801, 1884
Seong, Woojae–1883, 1936
Sereno, Joan A.–1911
Shafer, Benjamin–1651, Chair Session 3aSAa (1791), Cochair Session 3pSA (1823), Cochair Session 4aSA (1880)
Shafiro, Valeriy–1876, 1909
Shah, Parthiv–1670
Shahsavarani, Somayeh–1716
Shamma, Shihab–1783
Shang, Dajing–1692
Shang, Dejiang–1692, 1693
Shapley, Robert M.–1759
Sharifai, Nima–1862
Sharma, Nitin–1858
Sharma, Sneha–1703
Sharp, Phoebe C.–1893
Shaw, Emma–1709
Shaw, Sarah A.–1901
Sheehan, Caroline–1798
Sheffield, Eric–1710
Sheft, Stanley–1876, 1909
Shehorn, James–1875, 1877
Shekhar, Akshay–1701
Shekhar, Himanshu–1745
Shen, Chen–1666, 1687, 1727, 1761, 1762, 1882
Shen, Yi–1684, 1871, 1876, 1898, 1909
Sheng, Zhiyu–1858
Shepherd, Micah R.–1794, 1823
Shi, Bing–1795
Shi, Rushen–1766
Shi, Yuanming–1932
Shin, Seulgi–1767
Shinn-Cunningham, Barbara–1717, 1722
Shofner, William–1720
Shore, Aimee–1872
Shport, Irina A.–1931, 1932
Shrivastav, Rahul–1797
Shrivastava, Shamit–1778
Shub, Daniel E.–1721
Shung, K. K.–1893
Sichlinger, Laura–1798
Siderius, Martin–1781
Sidorovskaia, Natalia–1855, 1856
Siebein, Gary W.–1695, 1738, 1739, 1918
Siebein, Gary–1695, 1918
Siebein, Keely–1695, 1738, 1918
Siegmann, William L.–1787, 1800
Sievert, Paul–1775
Sigal, Ian A.–1859
Sikdar, Siddartha–1811, 1921
Silbert, Noah H.–1685
Silveirinha, Mário G.–1792
Silverman, Ronald H.–1701
Sim, Leng–1662
Simmons, James A.–1737, 1743, 1744
Simon, Blake E.–1756
Simon, Julianna C.–1862, 1922, Cochair Session 5aBA (1920)
Simoni, Mary–1784
Simpson, Brian–1680, 1722, 1724, 1874, 1878, 1879
Simpson, Christopher–1660, 1870
Simpson, Jennifer M.–1663
Simpson, Michael–1717
Sinelnikov, Yegor–1883
Singer, Andrew–1883
Singh, Agoshpreet–1741
Singh, Ravinderjit–1906
Singh, Tania–1894
Sirlin, Claude B.–1921
Sit, Arthur J.–1675
Skaggs, Miranda–1878
Skinner, Jarrod–1903
Skinner, Kimberly G.–1871
Skoe, Erika–1820
Skorski, Edwin S.–1854
Slack, Charles M.–1755
Slezak, James M.–1733
Smallcomb, Molly–1862
Smargiassi, Andrea–1656
Smeulders, David–1768
Smiljanic, Rajka–1911, Cochair Session 2pID (1749)
Smith, Kevin B.–1708
Smith, Kieren H.–1806, 1918
Smith, Michael J.–1671
Smith, Nicholas A.–1730, 1731, 1764
Smith, Spencer–1877
Smith, Zane–1795
Smotherman, Michael–1775
Snively, Jonathan B.–1901
Soderstrom, Melanie–1728, 1729, 1763
Soepriatna, Arvin–1702
Soldati, Gino–1656
Soleymanpour, Mohammad–1931
Sollmann, Lea–1659
Solomon, Latasha–1662
Solway, Joe–1738
Sommerfeldt, Scott D.–1704, 1712, 1791, 1870, 1884, 1896
Sonder, Ingo–1869
Sonderegger, Morgan–1931
Song, Xindong–1785
Song, Yang–1737
Sottek, Roland–1898
Southee, Richard A.–1915
Souza, Pamela–1821, 1873, 1876, 1877
Spallino, William–1696

Sparrow, Victor–1707, 1714, 1885, 1899, 1903
Spencer, Caroline E.–1798
Spencer, Nathaniel–1722, 1724, 1879
Spofford, Chuck–1804
Spratt, Kyle S.–1924, Cochair Session 5aPA (1923)
Spratt, Kyle–1727
Srinivasan, Nirmal Kumar–1722, 1874, 1875
Srivastava, Rohan S.–1704
Stanton, Timothy K.–1653
Stavropoulos, Trevor–1722, 1877
Stearns, Aaron J.–1665, 1725
Stecker, G. Christopher–1720, 1759, 1871, Chair Session 2pPPb (1758)
Stecker, Julie M.–1720
Stein, Alejandra–1763
Stepp, Mary Ann–1894
Sterling, John A.–1824
Stern, Richard M.–1759
Stewart, Jeremy–1784
Stewart, Michael–1816
Stills, Robert–1709
Stilp, Christian–1705, 1788, 1822, 1827, 1909, Cochair Session 3aPP (1788), Cochair Session 3pPPb (1821)
Stobbe, David–1791
Stocker, Michael A.–1855, 1919, Chair Session 4aAB (1855)
Stoner, Michael–1858
Story, Brad H.–1912
Strelcyk, Olaf–1877
Strickland, Elizabeth A.–1718, 1878, Chair Session 2pPPa (1757)
Stromswold, Karin–1910
Strong, John T.–1739, 1780
Strong, William–1704
Stuart-Smith, Jane–1931
Stump, Brian W.–1660
Sturdy, Christopher B.–1807
Sturtevant, Blake–1925
Su, Haili–1703
Su, Hui–1814
Su, Xiaoqian–1745
Su, Xiaoshi–1727
Suarez Castellanos, Ivan–1894
Sü Gül, Zühre–1890
Sui, Ni–1726
Sukovich, Jonathan R.–1862, 1925
Sullivan, Kent–1741
Summers, Jason E.–1672, 1920
Summers, Robert J.–1872, 1909
Sumner, Meghan–1689
Sundheimer, Alyxandria–1720
Supin, Alexander–1719, 1776
Supponen, Outi–1892
Surappa, Sushruta–1881
Sutlive, Joseph–1741, 1742, 1744, Cochair Session 3aABb (1776)
Svirsky, Mario–1766
Swallow, John C.–1697
Swaminathan, Jayaganesh–1720
Swayne, Billy–1680
Swearingen, Michelle E.–1661, 1681, 1694, 1870, 1900, 1901, Cochair Session 3aCA (1780)
Swift, S H.–1886
Swift, S. Hales–Cochair Session 4pNS (1897)
Sysueva, Evgeniya–1776
Szuberla, Curt–1868
Tabak, Gizem–1883
Taft, Benjamin N.–1826
Tahamtan, Mahdi–1929
Tai, Yihsin–1716, 1907
Taite, Winter–1933
Tak, Shubha–1922
Takahashi, Chikako–1826
Talebzadeh, Arezoo–1658
Talmadge, Carrick L.–1867
Tam, Alex–1861
Tamati, Terrin N.–1796, 1821, Chair Session 3aSC (1794)
Tan, Roderick–1859
Tan, Tsu Wei–1671, 1770
Tancredi, Gonzalo–1903
Tang, Dajun–1803
Tang, Rui–1771
Tao, Sha–1828, 1934
Tarakanov, Mikhail–1776
Tarr, Eric W.–1877
Tasko, Stephen M.–1816
Tatoglu, Akin–1733
Taubitz, Sarah–1711
Taylor, Caroline–1764
Tazawa, Makio–1671
Tennakoon, Sumudu P.–1925
Terpstra, Toby–1890
Thacker, Jared W.–1677
Themann, Christa L.–1866
Theodore, Rachel M.–1688, 1796, 1913, Cochair Session 1pSC (1688)
Thery, David–1890
Thomas, Erik R.–1931
Thomas, Madeleine–1909
Thomas, Reju G.–1745, 1892
Thompson, Charles–1733, 1817, 1864, 1902, Cochair Session 3aPA (1786), Cochair Session 3pPA (1817)
Thompson, Eric R.–1680, 1722, 1724, 1879
Thompson, Stephen C.–1895
Thompson, William Forde–1813
Thomson, Scott L.–1704
Thorson, Jill C.–1798
Thotala, Dinesh–1862
Tian, Zhenhua–1666
Tiede, Mark–1796
Tiemann, Christopher–1856
Tin, Jessica–1822, 1915
Tipton, Nathan D.–1903
Titovich, Alexey S.–Cochair Session 2aSA (1725), Cochair Session 2pSA (1760)
Titze, Ingo R.–1709
To, W. M.–1753, 1918
Tobin, Stephen J.–1911
Todd, Bryan D.–1700, 1864
Todd, Eric–1816
Tollefsen, Dag–1801, 1886
Tollin, Daniel J.–1717
Töpken, Stephan–1897
Torti, Valeria–1919
Tortoli, Piero–1892
Toscano, Joseph C.–1763, 1789, 1878
Tournat, Vincent–1705, 1754
Townsend, Simon–1766
Trader, Jason M.–1920
Trahiotis, Constantine–1684
Transtrum, Mark K.–1816, 1936
Treweek, Benjamin C.–1924
Tripathy, Aditi–1671
Trivett, David–1937
Trosclair, Tyler–1733
Trostel, John–1902
Tsaih, Lucky S.–1738
Tsysar, Sergey A.–1857
Tu, Jung-Yueh–1913
Tucker, Benjamin V.–1912, 1934, Cochair Session 2pED (1748)
Tucker, Jeff–1732
Tufariello, Justin–1661
Tufte, Steve–1751
Turgut, Altan–1803
Turner, Joseph A.–1706
Twitty, Ava B.–1681
Tyler, Jeramey–1814
Tzeng, Christina–1689
Ulrich, Micah–1860
Ulrich, Timothy J.–1755
Underwood, Samuel H.–1806
Undurraga, Eduardo A.–1784
Upadhyay, Awaneesh–1892
Urban, Matthew W.–1860, 1861
Uzhansky, Ernest–1671
Vakakis, Alex–1725
Vakili, Ali–1810
Valdes, Joaquin–1784
Valente, Daria–1919
Valentine, Greg A.–1869
Vallier, Timothy J.–1730, 1764
VanDam, Mark–1729, 1763, 1764, 1765, Cochair Session 2aSCa (1728), Cochair Session 2aSCb (1730), Cochair Session 2pSC (1763)
van de Par, Steven–1897
van der Steen, Antonius F. W.–1778
Vanegas, Jairo–1817
Van Emmerik, Richard–1872
Van Engen, Kristin–1827
Van horn, Eric–1896
Van Komen, David F.–1935, 1936, Chair Session 5aUW (1935)
van Sloun, Ruud J.–1674
Van Soest, Gijs–1778
Van Uffelen, Lora J.–1707
Vasandani, Veera S.–1789
Vasil, Kara–1876
Vayur, Kritika–1804
Vendittis, D. J.– Session (1734)Venegas, Gabriel R.–1672
Venezia, Jonathan–1685
Verhagen Metman, Leonard A.–1797
Verhey, Jesko L.–1793
Verhulst, Sarah–Cochair Session 4pPP (1904)
Vetterick, Matthew–1695, 1738, 1739, 1918
Vidrine, Lauren V.–1932
Vigeant, Michelle–1899
Vigeant, Michelle C.–1900
Vigness-Raposa, Kathleen J.–1671
Vignola, Joseph F.–1824
Villard, Sarah–1737
Vilov, Sergey–1779
Viswanathan, Vibha–1717
Vorlaender, Michael–1853, 1889
Vos, Hendrik J.–1892
Vuyk, Peter–1823
Wade, Lucas–1903
Wadsworth, Spencer–1816
Wage, Kathleen E.–1732, 1733, Chair Session 2aSPa (1732)
Wagner, Caleb M.–1662
Wagner, Ethan–1884
Walber, C.– Session (1734)Walczak, Mateusz–1701
Walker, Brandon–1742
Walker, Chad–1876
Walker, Elizabeth–1879
Wall, Alan T.–1662, 1680
Wallen, Sam–Chair Session 4pPAb (1902)
Wallen, Samuel P.–1687, 1726, 1754
Walsh, Edward J.–1806
Walsh, Timothy F.–1727
Wamsley, James–1928
Wanek, Austin M.–1703
Wang, Chaohui–1818
Wang, Dayu–1693
Wang, DeLiang–1874
Wang, Emily–1797
Wang, Guan–1667
Wang, Jiaheng–1937
Wang, Jianghua–1659
Wang, Lihong–1858
Wang, Lijun–1667
Wang, Lily M.–1696, 1706, 1806, 1918, Session (1829)
Wang, Michelle–1722
Wang, Michelle H.–1874
Wang, Ruihao–1744, Cochair Session 3aABb (1776)
Wang, Shunjie–1927
Wang, Xi–1795
Wang, Xiaoqin–1785
Wang, Xiuming–1768
Wang, Xiwei–1805
Wang, Xueding–1810
Wang, Yanhao–1741
Wang, Yingxiao–1893
Wang, Yuanyuan–1690, 1729, 1766
Wang, Yuqi–1891
Wang, Yuxiang–1664
Ward, Arthur–1923
Wark, David J.–1821
Warlaumont, Anne–1729, 1763
Warner, Emilee–1704
Warner, Natasha L.–1912, 1915
Warnick, Sean–1816
Watson, Charles S.–1683, 1795, 1821
Watts, Alison–1654
Waxler, Roger M.–1867, 1868, Cochair Session 4aPA (1867), Cochair Session 4pPAa (1900)
Way, Evelyn–1697
Wayland, Ratree–1825, 1929
Wazaefi, Khaled–1854
Weaver, Richard–1756, 1762, 1791, 1903
Webb, Jeremy–1893
Weber, Jonathan R.–1657

Weber, Thomas C.–1653, 1692, 1706
Weidner, Elizabeth F.–1653
Weikel, Stephen–1802
Weinandy, Rene–1679
Wells, Nathaniel–1896
Wen, Haiqi–1719
Wesolowsky, Michael J.–1697
West, Elllington–1923
West, James E.–1923
Westermann, Adam–1873
White, Andrew–1670
White, Michael J.–1661, 1681
White, Rebekah–1781
Whiteford, Kelly L.–1719, 1749, Cochair Session 2aPPa (1716), Cochair Session 3pPPa (1819)
Whiting, Jennifer–1889
Whitmal, Nathaniel A.–1914
Wicker, Ryan–1876
Wiebe, Peter–1653
Wiederhold, Curtis–1727
Wiley, Brandon M.–1673
Willardson, Matthew L.–1715
Willey, Carson–1686
Willi, Megan M.–1912
Williams, Grace–1877
Willis, John–1733
Wilson, Chad–1925
Wilson, Colin–1689
Wilson, D. K.–1800, 1812, 1885, 1900
Wilson, D. Keith–1661, Cochair Session 4aUW (1885)
Wilson, Preston S.–1672, 1707, 1727, 1807, 1811, 1887
Wince, Jess–1875
Winker, D. F.–Vice Chair, ASC S12 Session (1773)
Winn, Matthew–1760, 1790, Cochair Session 3aPP (1788), Cochair Session 3pPPb (1821)
Wiseman, Luke M.–1860
Wiskin, James–1857
Wisniewski, Matthew–1878
Wixom, Andrew S.–1725, 1887
Woeste, Hannah M.–1794, 1799
Wojtczak, Magdalena–1875
Wolfenstein, Kiley–1764
Wolfram, Frank–1656
Wong, Kainam T.–1737, Chair Session 2aSPb (1737)
Wong, Kainam Thomas–1801
Wong, Lena–1879
Woodruff, Steven R.–1669
Woolard, Alix–1767
Woolfe, Katherine F.–1804
Woolworth, David S.–1696, 1854
Woolworth, David–Cochair Session 2aAAb (1697)
Worcester, Peter F.–1803
Worland, Randy–1750, Chair Session 2aMU (1708)
Worthmann, Brian M.–1804
Wright, Hannah M.–1877
Wright, Richard–1877
Wu, Kang–1667
Wu, Lixin–1771
Wu, Min–1778
Wu, Rui–1896
Wu, Yashuo–1763
Wu, Yu-Hsiang–1717
Wu, Yushuang–1693
Wyles, Saranya P.–1676
Xiang, Liangzhong–1920
Xiang, Ning–1658, 1708, 1802, 1805, 1806, 1824, 1888, Cochair Session 3aSP (1800), Cochair Session 4pAA (1888)
Xiang, Zhihai–1823
Xiangyang, Zeng–1888
Xiao, Yan–1693
Xie, Hua–1922
Xie, Xin–1689
Xu, Can–1828, 1934
Xu, Katherine–1765
Xu, Stephany Y.–1867
Xu, Zhen–1862, 1925
Xue, Xiaoyan–1805
Yamada, Tsuneo–1826
Yamaguchi, Tadashi–1859
Yamamoto, Hiroaki–1768
Yan, Xiang–1659, 1726, 1805
Yang, Bin–1859
Yang, Bingen–1750
Yang, Haesang–1883
Yang, Luhui–1698
Yang, Meng–1912
Yang, Ming–1753
Yang, Rui–1924
Yang, Xinmai–1810
Yang, Yike–1825, 1929
Yang, Zhengwei–1882
Yao, Jiamin–1667
Yarina, Marina–1671
Yathiraj, Asha–1922
Ye, Yi–1905
Yengul, Sanjay S.–1858
Yin, Anorexia–1699
Yin, Heng–1795
Yin, Rong–1913
Yin, Xiaoyan–1699
Yiu, Billy Y.–1701, 1703
Yiu, Suki–1911
Yoas, Zachary W.–1794
Yoon, Sangpil–1893, Cochair Session 4pBAb (1892)
Yoritomo, John Y.–1756, 1762, 1903
Yoritomo, John–1791
Yost, William–1685, 1723, 1783, 1873
Young, Jonathan D.–1880
Young, Sarah M.–1755
Yousefian, Omid–1781, 1860, 1861, 1893
Yu, Alfred C.–1701, 1703, Cochair Session 5aBA (1920)
Yu, Jaesok–1703, 1859
Yu, Lan–1828
Yu, Tzu-Ling J.–1790
Yu, Xinyue–1771
Yuan, Fuh-Gwo–1726
Yuldashev, Petr V.–1902
Yun, David–1876
Yun, Donghyeon–1871, 1876
Yusupov, Kamil–1769
Zabolotskaya, Evgenia A.–1715
Zabotin, Nikolay A.–1869
Zabotina, Liudmila–1869
Zahorik, Pavel–1705, 1875, 1877
Zaman, Khaleela–1658
Zanoli, Anna–1919
Zarafshani, Ali–1920
Zderic, Vesna–1894
Zeh, Matthew C.–1887, Cochair Session 2aID (1704)
Zelem, Lukas–1888
Zeng, Yuyu–1911
Zhai, Yuxin–1687
Zhang, Chao–1692, 1693
Zhang, Chengwei–1769
Zhang, Dongdong–1915
Zhang, Jie–1911
Zhang, Jun–1937
Zhang, Junjian–1781
Zhang, Likun–1655, 1706, 1787, 1804, 1817, 1818
Zhang, Liujun–1698
Zhang, Qianchu–1670
Zhang, Shuxin–1699
Zhang, Weifeng G.–1886
Zhang, Xiaoliang–1732
Zhang, Xiaoming–1673, 1675, 1676, Cochair Session 1aBAb (1656), Cochair Session 1pBA (1673)
Zhang, Xiaoqing–1881
Zhang, Xin–1726, 1896
Zhang, Xuedong–1771, 1935
Zhang, Yang–1764
Zhang, Yehui–1802
Zhang, Yingzhen N.–1921
Zhang, Zhuohuang–1876, 1898
Zhao, Anbang–1693, 1732
Zhao, Han–1937
Zhao, Ningxiner–1669
Zhao, Yan–1874
Zhao, Zhigen–1859
Zhargarian, Amir–1903
Zheng, Bin–1920
Zheng, Qi–1731
Zheng, TengFei–1818
Zheng, Xiaoyu–1761
Zheng, Xiaoyu (Rayne)–1665
Zheng, Z. C.–1781
Zheng, Zheng–1802
Zhou, Boran–1673, 1675, 1676
Zhou, Helen–1814
Zhou, Jinling–1673
Zhou, Yi–1906
Zhu, Hongfei–1670
Zhu, Jie–1881
Zhu, Jinying–1706
Zhu, Lifei–1862
Zhu, Xiaohui–1687
Zilany, M. S. A.–1664
Zimmerman, Richard–1793
Zimpfer, Veronique–1661
Zinck, Scott A.–1922
Zinoviev, Alex–1770
Zou, Chengzhe–1669
Zou, Huang–1814
Zou, Zheguang–1655
Zurek, Patrick–1721

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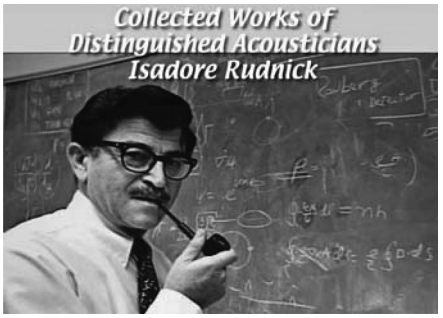
Collected Works of Distinguished Acousticians

Isadore Rudnick



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

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

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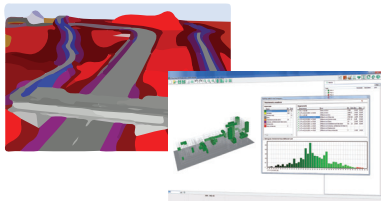

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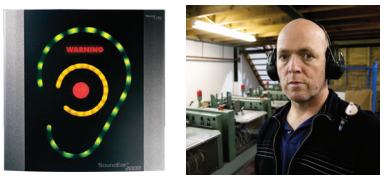

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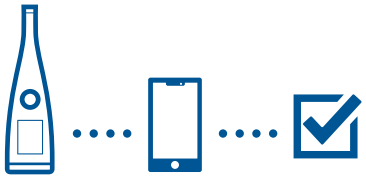

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